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THE DEVELOPMENT OF A LOW-COST SYNCHRONIZED PCM DIGITAL AUDIO SYSTEM FOR VIDEO PRODUCTION

David W. Kelln

Music Department, McGill University

towards a Master of Music (M.Mus.) degree

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ABSTRACT

Due to the inadequate noise, distortion, and frequency response characteristics of the analog audio tracks found on video tape recorders, the creative potential of this side of a video production is extremely limited. The development of low-cost pulse code modulation digital audio processors and time code-based synchronizers offers a means to bypass this limitation and achieve very high-quality audio response.

Described in this report are tests which indicate the feasibility of using these components in conjunction with a programmable video editing system to record, edit, and replay digital audio in synchronization with a video image. A description is also given of the first complete video production using this system. This details the recording of synchronized digital audio and all aspects of post-production, including interconnection wiring, use of a synchronized multitrack analog recorder, layback to digital audio, and subsequent editing onto the digital audio master of the production.

MISE AU POINT D'UN SYSTÈME AUDIO DIGITAL PCM SYNCHRONISÉ À FAIBLE CÔUT POUR DES PRODUCTIONS VIDÉO

DESCRIPTIF

A cause du bruit de fond, de la distorsion et des caractéristiques de réponse de fréquence des bandes audio similaires sur le marché des magnétophones vidéo, le potentiel de création dans cette gamme de production vidéo est très limité. La mise au point de processeurs audio digitaux à modulation de code à faible côut et de synchronisateurs de cadence offre un moyen de contourner ce problème et de réalise une réponse audio de très haute qualité.

Des tests sond décrits dans ce rapport qui indiquent qu'il est possible d'utiliser ces composants avec un système d'édition vidéo programmable pour enregistrer, éditer et repasser un audio digital synchronisé avec une image vidéo. Nous donnons également une description de la première production vidéo complète utilisant ce système. Nous donnons également les détails de l'enregistrement d'un audio digital synchronisé et de tous les aspects de post-production, y compris le système de branchement électrique, l'utilisation d'un magnétophone similaire synchronisé multi-pistes, le retour à un audio digital et l'édition en résultant sur la matrice audio digitale de la production.

A NOTE ON TERMS USED IN THIS PAPER

The following is a list of terms and abbreviations used in this paper. These are used in their abbreviated form in the body of this paper because they are the standard terminology used in the audio and video engineering and production fields. Also included is a list of registered trademarks for those pieces of equipment (and the manufacturers thereof) used in this project and referred to extensively in this paper. Notice of trademark should be assumed throughout the body of the text.

A.C.	alternating current
ATR	audio tape recorder
B.B.C.	British Broadcasting Corporation
BVU	broadcast video unit
C.B.C.	Canadian Broadcasting Corporation
dB	deciBel
D.C.	direct current
E.I.A.J.	Electronics Industries Association of Japan
F.S.K.	frequency shift keying
Hz (kHz)	Hertz (kiloHertz)
ips	inches per cecond
NTSC	National Television Systems Committee
P.A.	public address
PCM	pulse code modulation
PZM	pressure zone microphone
RMS	root-mean-square
SMPTE	Society of Motion Picture and Television Engineers
S/N	signal to noise ratio
VCR	video cassette recorder
VTR	video tape recorder
VU	volume unit

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Ampex is a trademark of Ampex Corporation.
Commercial Electronics is a trademark of Commercial Electronics Ltd.
dbx is a trademark of dbx, Incorporated.
Dolby is a trademark of Dolby Larboratories Licensing Corporation.
JVC is a trademark of JVC Electronics of Canada, Ltd.
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ABBREVIATIONS AND TRADEMARKS continued

Mitsubishi is a trademark of Mitsubishi Electric Corporation.
Omni Q is a trademark of Commercial Electronics Ltd.
Sony is a trademark of Sony Corporation.
Studer is a trademark of Studer-Revox Incorporated.
Teac is a trademark of Teac Corporation of America.
Telcom is a trademark of Telefunken AE-Gevaert Incorporated.

TABLE OF CONTENTS

PREFACE

The work described in this document is totally original in that, though it involves the application of existing technology, it had not, to the best of my knowledge, ever been done before. The video production described in this thesis, J'Aurais Dit--Glenn Gould, is the first in Canada to be made with a parallel digital audio master as its sound track. Since completion of this project, I have found reference in a magazine article to a similar application of digital audio to video production involving video and digital audio recording of a performance of the Metropolitan Opera in New York.[1]

The concept for this system, and belief in its feasibility, came from previous personal experience and exposure to audio engineering, the use of video tape recorder-based PCM digital processors, video editing, post-production audio work for video productions, and research into the means by which digital audio can be stored on magnetic tape.

The impetus to pursue this project came from experiences in post-production audio at a low-budget but quite technically sound facility at Western Front, Vancouver, British Columbia. The majority of the video productions undertaken at this facility fall into the category of video art works. This work, which was exploring new aesthetics and techniques in the video domain, was extremely limited in its audio possibilities by the capabilities of the equipment, which consisted of professional broadcast-quality 3/4-inch video recorders.

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[1] Bob Katz, "Getting Started in Digital Audio Recording and Post-Production," Recording Engineer/Producer, April 1984.

The use of dbx noise reduction was explored to attempt to achieve some useable dynamic range. However, problems such as poor frequency response and self-generated noise caused very evident mistracking and audible 'pumping' of background noise. Use of more sophisticated noise reduction systems may have worked better, but at a much higher cost and with no solution to the frequency response problem. The only sure avenue of escape from the dilemma was use of a double system; that is, a separate high-quality audio recorder synchronized in some manner to the video recorder. This technique has been standard procedure in the film industry for many years both for location and post-production audio.

My previous work with video drive-based PCM digital audio processors indicated to me that this was the most cost-effective means of achieving excellent audio quality--though with perhaps some limitations in operational capabilities. This work included use of the Technics SV P-100 for replay of test material for subjective measurement experiments on loudspeakers while acting as research assistant to Dr. F.E. Toole at the Acoustics Section--Physics of the National Research Council in Ottawa, Ontario, and participation as research assistant in subjective comparisons of digital processors including the above Technics and the Sony PCM-F1 with high-quality analog audio mastering recorders. This work was the project of Dr. Wieslaw Woszczyk of McGill University and Dr. F.E. Toole of the National Research Council. The results of this research indicated that the technical and sonic quality of most of the digital processors tested was comparable to the analog recorders which were equipped with noise reduction, though at a much lower cost.

The use of synchronizers (usually based on the SMPTE standard) to lock various combinations of audio and video recorders together is quite common. Though I had had no previous experience with such a device, it seemed that it should work equally well if the material on one of the video recorders happened to be video format digital audio.

The limitation that could possibly obviate the use of the lower-cost video-based PCM digital audio processors was the lack of a proper editing control unit that could perform the data processing that should occur at edit points. If editing capability could be established with a standard video editing system, the way would definitely be open to the application of a parallel digital audio system to video production. A further capability that it was necessary to establish was the ability to do audio and video edits at different times using the same video editing system. If by use of the time code as a guide it was possible to make edits like this to a close enough tolerance for the application, an actual production could be attempted. This would require use of all the audio post-production techniques commonly employed, be proof of the system's capability, and undoubtedly uncover necessary operational techniques and limitations, if any.

Before any of this could commence, however, some development work was required on the synchronizer that I had chosen. This was a very inexpensive device intended for audio recorder control and not built to operate in accordance with the SMPTE standard. This device is the Omni Q TL1 synchronizer, which is manufactured by Commercial

Electronics Limited, Vancouver, British Columbia. It was immediately obvious when discussing the problem with engineer Mr. Mark Yau that the Omni Q did not have the circuitry necessary to control a video recorder in the conventional way. On my suggestion, Mr. Yau constructed an interface module for the synchronizer which produced a variable frequency pulse wave output with a center frequency of approximately 60 Hz and of polarity and voltage levels to be compatible with the External Playback Sync input found the Sony BVU-200B video recorders which were to be used. This input is normally used to maintain constant speed and frame relationship of the video recorder on playback by referencing it to an external frequency reference at the correct video frame rate. This did, however, function quite well at varying the speed of the recorder enough to bring it into sync in a reasonable length of time if both master and slave drives were reasonably close to the desired relationship when started.

Apart from the work on the Omni Q interface for the video recorder and an additional interface which enabled it to also control the Teac model 38 eight-track recorder which was part of the Western Front audio facility, all work on the development of this system and the operational procedures for its effective use was done by myself exclusively, and all contents of this thesis with the exception of excerpts from various manuals for the equipment employed, which are found in the appendices and so noted, were authored by myself.

I would like to acknowledge the assistance of Ann Hepper, Ruth Hibberson, and Clark Steabner for their contribution as performers to

provide material for the initial tests of the system. I would also like to thank Ann Hepper for proof-reading and typing the first draft of this work, and Douglas Brown for typing the final copy.

The video production which is described is entitled J'Aurais Dit-- Glenn Gould, and is the creative work of artist Rober Racine. M. Racine was invited by the video director of the Western Front, Kate Craig, to be artist-in-residence at this facility and do this production. All original audio recording and all post-production audio for this work was performed by myself. Those audio materials brought by M. Racine to be used in the production (interviews, commercial recordings, excerpts from Canadian Broadcasting Corporation radio plays) are noted in the detailed description of the work on this production.

Finally, I would like to thank Mr. Fred Lichota from Sony Corporation of Canada, Richmond, British Columbia, for arranging for the use of a Sony PCM-F1 processor and his encouragement and interest. I would also like to thank the gentlemen at Commercial Electronics Limited, Vancouver, British Columbia, for the use of their Omni Q synchronizer, and the Integrated Media section of the Canada Council for the Arts which provided the research grant to cover the many costs which a project such as this incurs.

Some discussion is in order here about the approach that has been used in this thesis. I have endeavoured here to produce a manual that will allow others of perhaps varying backgrounds to set up and effectively operate a similar system. A requirement was that it should be accessible to people working in both audio and video without necessarily having much exposure to the other discipline. Further requirements were that it should not require one to have the background knowledge of a professional engineer or technician in either of these fields, and the system itself should not require a large budget to implement.

The constraints mentioned came from the facility that this system was developed for, and the funding agency for this research. The facility was Western Front in Vancouver, British Columbia, which is primarily a production centre for video art productions. It is representative of a number of similar facilities across Canada which provide production resources to video artists. The funding agency for this project was the Integrated Media section of the Canada Council, which was interested in a practical application which could provide additional resources to these facilities.

As a result, this presentation may tend to be simplistic in areas that the reader may already be familiar with, and also may require careful reading when referring to aspects that the reader is unfamiliar with. In all the technical areas that are a part of this project (digital audio, video editing, synchronization, etcetera) I have endeavoured to include in the appendices the necessary background information in the form of manuals for the equipment used, interconnection wiring diagrams for this system, etcetera. This should help to provide additional information to those unfamiliar with a type of equipment or the particular model of equipment that was used. These appendices, combined with the discussion of the theory, the testing, and the use of this system as described in the body of this work, should also enable one to realize a similar system without requiring exactly the same equipment.

Due to the exploratory nature of this project, there was also some difficulty with providing a good list of bibliographic references. Those articles that I have found that deal specifically with the application of digital audio to video production are included. The majority, however, deal with only a particular aspect of the varying technologies used. As with the appendices, these are to provide background information on aspects of the technologies that the reader may be unfamiliar with and which cannot practically be included in detail in this work.

I. INTRODUCTION

The first thing to say in regard to the title of this report is that it is possible. I'm happy to start with this, since it's far less satisfying to prove that something can't be done. I must add, however, that this can only conclusively be said for the equipment we were using. While we did generally have good equipment for this project, it was not the most modern or sophisticated. I have confidence that anyone with a reasonable technical acumen can realize use of a system such as this. Where any specific incompatibilities of equipment are known, they are noted in the various parts of this report.

An important aspect of how this project was approached was the cost involved. Since the Western Front is a member of the Association of National Non-Profit Artists' Centres (ANNPAC), the budget is not large--especially for audio. The question was, "Can an affordable digital audio system be set up that will work for video production?" The answer is "yes," if:

--you know something about the equipment and how it functions; and

--you are willing to engage in a fairly rigorous production and post-production procedure.

The problem with 'affordable' is that it usually means that you sacrifice ease of operation (so it is more labour intensive), and user friendliness (so you must know the system, sometimes adapting yourself to its weak points). The aim of this report is to detail my experience with a system such as this, and address these issues.

There must be one more proviso, which is that you must want the best audio quality possible and must know how to get it. All that the digital audio system will do for you, really, is get the recording medium out of your way. What you feed into the recording processor is what you get out. Attention must be paid to the whole recording chain at each point in the production.

II. DOUBLE SYSTEM AUDIO WITH IMAGE: HISTORICAL PERSPECTIVE

The use of a separate audio system to accompany an image medium has a history that goes back almost one hundred years. The motivation for this kind of system has come from two factors; one is that often the technology has not been in place to put audio directly onto the image medium and thus absolutely requires a separate system, the other is that often a system that allows the inclusion of the audio material on the image medium does not have very good audio quality. It is the latter situation that is the motivation for this project.

The initial double system work was done with film, as this medium preceded the development of video by a number of years. The first system could be said to be the piano player accompanying the silent film while watching it in the theatre. A more sophisticated situation was that used in D.W. Griffith's The Birth of a Nation in 1915, which used performers behind the screen delivering dialogue and creating sound effects. The first real system involving an audio reproduction medium with film was the use of recorded discs, which were played in synchronization with the film. Work by Thomas Edison and his assistant W.K. Laurie Dickson was undertaken which produced the Kinetophonograph, which was a combination of the Edison phonograph with a Kinetoscope (also his invention). This was first demonstrated in 1889.

It was as early as 1900 that the first patent for a system which recorded audio directly on film as an optical signal was filed. However, at this time there was no method of reliable electronic signal amplification to be used in conjunction with it. This lack prevented the dissemination of this technique for many years. It was in 1923 that inventor Lee DeForest had a system which he called Phonofilm, and which was acquired by the Fox film company and used in that studio's film process known as Movietone.

It was, however, the synchronized phonograph disc which was first used in commercially released films to begin the age of film sound and use of double systems. The movie company Warner Brothers, in conjunction with the Western Electric Company, developed a sound system referred to as Vitaphone which used synchronized discs. In late 1926 a demonstration film was released with this process, and in 1927 the film The Jazz Singer was released. This film is generally regarded as the first movie which launched the age of 'talking' films. The Vitaphone system did not last very long, though, due to the problems of keeping the audio discs and the film synchronized, and within a few years was completely replaced by optically-encoded sound directly on the film stock itself.

At first the sound-on-film technology was comparable to that achievable on the discs of that period, but by the time of the Second World War, film sound had become significantly inferior to the quality achievable on disc or on radio broadcast. The next development to overcome this fidelity problem came after the Second World War with the commercial development by the Ampex corporation of the magnetic tape recorder. This technology was developed by the Germans during the war, and a few working models were found after the war and were copied and developed by the Americans. This offered a significant improvement in quality by again returning to a double system, at least in the initial recording and production of a motion picture. The optical print remained the distribution medium for sound with film for many years.

As always when using a double system, the first thing that had to be done when developing the use of magnetic tape recording in conjunction with film shooting was to provide a means so that the recorder and camera image could be synchronized later. There were two aspects to consider: the first was to establish the initial correspondence between audio and image; the second was to ensure that the two playback machines ran at exactly the same speed and maintained that correspondence.

The initial correspondence was established by use of a 'clapper' board which had the information about the particular scene, take, etcetera, written on it, and was shot by the camera at the beginning of each take. This clapper board also had a hinged top part which was banged loudly against the bottom, giving an easily-established initial relation between the audio and the film. It was not attempted to control the audio recorder's speed duing the initial filming; instead, a time reference was recorded on a track of the audio recorder which could later be used to establish its speed in relation to the film. Typically, this reference was taken from the power line frequency on which the camera speed itself was dependent. This process has become known as 'pilot tone'. The stage following the initial shooting of a scene involves use of this recorded pilot tone to vary the speed of the audio playback unit while the audio is transferred to magnetic stock that is sprocketed. Once this was successfully accomplished, the synchronization of the audio and film depended on mechanical systems which locked the sprocket drives of the audio stock and the film stock together. Pilot tone has often been referred to as a system of 'invisible sprockets'. This system of sprocketed magnetic stock has developed into the systems now in use in film sound mixing theatres, where multiple units with two-inch sprocketed stock having six magnetic tracks are all synchronized together mechanically. This is then mixed to a similar unit which is also mechanically synced to all the other units, including the projector showing the film, while the mix occurs.

Once the master sound track was in finished form it then had to be transferred to a medium which can accompany the film to its place of showing. Initially this was only transferred to an optically-encoded audio track on the film. This was developed further with film stock that could be striped with one or more magnetic tape tracks. It is this medium that represents the highest standard today in film presentation. Often up to six magnetic tracks, Dolby noise reduction encoded, are used in a 'surround-sound' presentation system for major releases.

The development of television and video tape recorders brought about a similar through reversed development. Since the video recorders used magnetic tape to record the video, it was natural to use magnetic audio tracks directly on the video tape right at the beginning. Unfortunately, the type of magnetic tape (its oxide thickness and type) was optimized for the much higher frequencies of video signal recording. For example, the standard of the National Television Systems Committee (NTSC) which was adopted for North America requires a four megaHertz bandwidth. The requirements for recording this type of signal are quite different than for recording a one hundred Hertz signal. A further limitation that developed was the lack of a multitrack audio production format that could include the image. Even today the standard for broadcast one-inch video tape includes only three audio tracks, and they are not of the quality of a conventional audio recorder.

To overcome these limitations and make use of the rapid development of multitrack audio formats that were being developed for record production, another system of 'invisible sprockets' was developed. This took the form of a time code that could be recorded on both the video recorder and analog audio recorder. This time code, developed as an international standard by the Society of Motion Picture and Television Engineers (commonly referred to as SMPTE), could identify the time position of a program in hours, minutes, seconds, and frames. A device which could read this time code and determine the respective positions of the audio and video material could generate a control signal to ensure one machine would always stay in synchrony with the other. This code has come to be know as SMPTE longitudinal time code. As well as the use of synchronizing a separate audio recorder for production purposes of a sound track for video, SMPTE time code has also come to be used to automate video editing functions.

The present procedures which are used in the production of a video sound track include the initial recording of image and appropriate sound, transferring the initial sound to be used to a multitrack audio recorder using recorded time code to keep the synchronization of audio and image, adding on the multitrack recorder the necessary sound effects and music, and finally transferring the mixed sound track to the video master tape. This is often accomplished now with a special unit referred to as a 'layback' recorder. This recorder has all electronics and heads optimized for audio frequency signals, but can record on one-inch video tape. Thus, the audio and reference time code are transferred onto the tape that will eventually be the final master for the project. Usually the video image edit is complete before the production of the sound track takes place. A copy of this is then synchronized to the multitrack audio machine and is used during the production and mixing process of the sound track. The final mix of the sound track is then made to, or transferred to, the video master, or if using a layback machine a transfer of the video edit is made to the tape which has the sound track already on it.

We are now entering a period where digital recording technology is being used in the production process of audio for video and film. This development is especially needed in the area of video due to the development of distribution media such as the laser disc, which have the capability of having digital audio sound tracks. Also, it appears that it will not be too far into the future when we can expect to see (hear) digital audio broadcast of sound accompanying television. Work and tests have already been undertaken by the British Broadcasting Company in Great Britain in this regard.

The work described in this thesis is directed at exploring the use of this technology to improve audio quality available for production and distribution of video work, and at a price that will mean that this technology can be used by other than the major television networks and producers.

III. INTRODUCTION TO DIGITAL AUDIO

A. WHY USE IT?

One obvious initial question is, "Why go to all the trouble?" The answer is as follows:

	Sony F1	Sony BVU-200B audio
Signal/noise	86 dB at 0.01% THD	48 dB at 3% THD
Crosstalk	80 dB	(no specifications)
Wow & flutter	0.0	0.2% RMS
Frequency response	10 - 20 000 Hz at ± 0.5 dB	50 - 15 000 Hz
Distortion	0.007% (14 bit)	2.5% for 1 kHz at reference level

The basic problem is that the audio tracks on a broadcast VCR (even one-inch) have either as much or more noise than a compact cassette. I personally was tired of having to have all audio levels as close to saturation as possible at all times just to get acceptable sound later. With material that has a wide dynamic range, you are just out of luck.

In tests performed by the graduate recording faculty at McGill University, Montreal, and the Acoustics section of the National Research Council, Ottawa, it was found that the specifications and sonic results of linear PCM encoders matched those of high quality analog recorders with noise reduction; e.g. Studer A80 with Telcom noise reduction, and Ampex ATR-100 with Dolby A.* Considering that these professional analog

*Dr. F.E. Toole and Dr. W. Woszczyk, "A Subjective Comparison of Five Analog and Digital Tape Recorders," Audio Engineering Society Preprint #2033, 1983.

mastering decks with noise reduction will cost you at least $10 000 or more, there is obviously something worth investigating here. Another aspect is that in the ANNPAC system, those that would most likely be interested in this capability (those into video production and presentation) should already have the video equipment necessary to realize it.

The additional equipment necessary in the form of the synchronizer adds additional capabilities that can be extremely useful, such as use of a multitrack analog recorder synced to the video.

B. HOW IT WORKS

If you can measure the amplitude of a wave form at a high enough frequency (at least twice the highest audio frequency you want) and can convert that measurement from an analog voltage to a number within a certain limited range (for example 0 to 65 000), you can later reconstruct the wave form with no loss of information. All this tells us so far is that it is possible to record audio as a series of numbers rather than as a continuously changing (analog) parameter such as voltage, current, or magnetic flux. Why do this? Because the capability of analog tape has been pushed as far as is possible, and this requires having everything optimized to loading the tape with the signal. In video you have very rudimentary audio circuitry, inadequate heads and shielding, and are using tape optimized for a video signal. Basically, there is nothing better about digital as opposed to analog except that by storing a number you bypass the characteristics of the recording tape. Because we are then storing bits of binary data, the system will either work extremely well or not at all.

This brings us to why a video machine is used as the storage recorder. The signal-to-noise ratio and distortion characteristics are directly related to the size of the number range you have to use. Roughly, you get 6 dB of S/N ratio for each binary digit (or bit) in the word. For example, the E.I.A.J. standard the Sony PCM-F1 follows calls for a 14 bit word size. The gives a S/N ratio of (14 x 6 =) 84 dB for a word of 14 bits. To follow the rule of the sampling rate being at least twice the audio frequency bandwidth you need (20 000 Hz), this standard uses a rate of 44 056 Hz. This then gives us a large number of bits per second for each channel of signal (44 056 x 14 = 616 784), and so two channels require approximately 1 233 000 bits per second. So, in order to store the information, we need a machine with a frequency bandwidth that is larger than this, which is just what a video drive has to have to be able to store a video signal. All that is necessary is to put the data into an NTSC format (with the vertical and horizontal pulses the video drive needs).

This is what the digital processor does:

--takes the analog signals in;

--converts them to 14 bit binary numbers at a rate of 44 056 times a second for each channel;

--adds some error detection and correction information;

--mixes the data up in a very particular way (interleaving), so tape dropouts don't wipe out a bunch of data for a particular time interval; and

--stores the numbers (made up of 1's and 0's only!) as levels of luminance information (white or black) within the NTSC video signal it generates.

When playing back, the processor simply reverses this procedure.

The encoding system that this E.I.A.J. standard uses is called linear Pulse Code Modulation (or PCM, as we will call it in this report). The system operates linearly insofar as the conversion from analog to numerical value follows a one-to-one correspondence. Pulse code modulation stands for the constantly clocked sampling of the analog voltage, and conversion to numerical code.

C. DIGITAL AUDIO AND VIDEO INTERFACE--EQUIPMENT REQUIREMENTS

The fundamental concept involved in realizing the use of digital audio in video production is a simple one. Because the digital audio information requres the complete video track of one VCR and the picture requires a complete video track of its own, we obviously need two VCRs; one for picture, and one for sound. These two machines will therefore have to be locked together in some manner, so that sound and picture start together and maintain their time relationship through the whole program when replayed.

The only way this can be conveniently accomplished is with a synchronization system using a time code recorded on an audio or SMPTE track of each VCR. With a time code there are the distinct advantages of having an absolute time indication with a resolution of one frame--you always know where you are in the program, can easily start both machines at any selected point from the beginning to end, and you can count on the machines running in very close sync once they are running together. One other important capability that time code gives is the ability to edit picture and sound at separate times and still maintain synchronization.

IV. PRODUCTION PROCEDURES--INITIAL CONCEPT

A. SHOOTING

Here is the initial concept evolved for use of a system such as this in an actual production. When shooting the rushes for the program, video and audio VCRs are put into record simultaneously. An identical time code signal is recorded on an audio track of each VCR while the program material is being recorded. Doing this will make the syncing easy in post-production, since the two time codes will be identical for each point in time. While shooting, the other audio track on both the video and PCM audio VCRs will be used to record a mono mix of the audio material. This gives you something to listen to when dealing with the video signal later, and acts as a guide to determine edit points on the PCM audio tape. One limitation of the PCM signal is that it can only be played back at real time speed. Therefore, the analog guide on the tape is essential to determine edit points accurately using slow scanning speeds.

B. EDITING

When editing, because we only have two VCRs in the editing system, the video and audio material have to be edited separately. Usually one of the media will act as the main reference. Quite often this will be the image, but in something like a musical production it would be the audio. The procedure then involves:

1) determining at what time code values the edits should take take place for the rush and master tapes;
2) doing the first edit (audio or video);
3) making the matching edit on the other medium's tapes (rush and master) at the same time code values.

By this method it should be possible to do a continuous series of edits that will later stay synchronized. Editing of the PCM signal will take place as a video 'insert' edit from an original PCM recording to the PCM audio master in the same manner as video image edits. Video 'insert' editing will be used exclusively because of the greatly reduced disruption in the video signal as opposed to 'assemble' editing (please refer to Appendix III: Excerpts from the Service Manual for the Sony BVU-200B Videocassette Recorder for the differences between 'insert' and 'assemble' edits).

V. SYSTEMS TESTS

This section will describe the initial tests before the first complete video production accomplished with this system. The production was J'Aurais Dit--Glenn Gould, by Rober Racine.

The discoveries made while making these tests form the basis of the procedures used in the video production itself. These descriptions and Appendix I--Equipment Interconnection and Interaction therefore serve as a main reference to the production procedures described in Section VI Video Production--J'Aurais Dit--Glenn Gould.

A. TEST #1--METRONOME

1. Description

To keep things simple, an electronic metronome that had a simultaneous click and flashing light was used as a test signal source.

The procedure was as follows:

--a camera was focused on the metronome;

--a microphone was set up to feed the PCM audio processor.

The equipment was set up as in Diagram 1: Recording Audio and Video Material. Notice that no interconnection for video sync was tried in this set-up. Both audio and video were completely separate systems. The only link between them was the time code being generated by the Omni Q synchronizer and being recorded simultaneously on audio channel 1 of both VCRs.

--Both video machines were recording in Assmeble mode utilizing the BVU editor as a remote control;

--the dropout compensator was turned off on the deck recording the PCM signal, as per manufacturer's instructions;

--on the PCM VCR, the framing servo was left on;

--a twenty minute recording of the metronome was made.

At the end of this time, both tapes were rewound and the synchronizer was set to read the time code from both decks, with the speed of one of the VCRs being controlled by the synchronizer. The synchronizer does this by generating a signal compatible with the External Playback Sync input on the BVU-200B VCR. This signal is a pulse wave of a certain frequency. The External Sync input works by comparing the frequency of the incoming signal to that of the servo pulses coming off of the tape, then varying the motor's speed so that they match. If the slave is lagging behind the master, the synchronizer would increase the frequency of the signal at the External Sync input to speed the slave up. Conversely, if the slave was leading the master, the synchronizer would decrease the frequency of the controlling signal to slow the slave. When both are in exact sync, the signal is the normal NTSC video field rate of approximately 60 Hz.

There is one limitation on which decks can be reliably used for PCM signal storage; it must be possible to disable any video signal dropout compensation! This is accomplished on the back panel of the BVU-200Bs, and there is a special cord connector provided with the Sony F1 that does this when it is used with the SL2000 portable 1/2" Beta-format VCR. I don't know if this connector will accomplish the same thing with any other deck. Note: U-matic machines may be a problem in this regard because they have an internal circuit that can't be turned off, and may need to be modified. There are some Sony Beta-format VCRs that have a PCM switch on them which will accomplish this; e.g. Sony SL2300, and Beta Hi-Fi decks.

2. Findings

1) You have to have a synchronizer.

This is not so much for the speed variation of the decks as these were quite constant, and when connected to the editor, there seems to be some control in operation. In fact, the editor must provide some control or insert edits would be impossible. However, it is impossible to start two decks simultaneously! The closest I came was, on average, about ten frames off, and once running, there is no way of varying the speed of the decks.

2) When connected to a remote control editor, the synchronizer would only work when the deck connected to the Player remote was the master and the deck connected to the Recorder was the slave. If this was reversed, the system acted as if two things were controlling the system and fighting each other, causing either a fluctuation in picture or muting of the PCM audio. This was, in fact, happening. The synchronizer was varying the speed of the Player in relation to the Recorder, while the editor/controller did the reverse; a feedback loop of no benefit at all.

3) The tracking of the unit playing back the PCM signal had to be adjusted before the signal could be retrieved. This was more evident on one VCR than the other, and may have to do with some aspect of the chroma component of the signal, which should only be black and white.

The following procedure was found to be the best for getting the machines running in sync initially when playing back. The Omni Q TL2 display unit gives you the option of displaying either:

--the master time code playback,

--the slave time code playback,

--the difference between these two, or

--the clock value being generated.

To start both machines in synchronized playback, I did the following:

1. I played the slave to a point about five seconds into the time code and then put it in Pause mode.
2. I put the master into playback and set the display to read the difference between master and slave time code values.
3. The display then showed the difference between the time codes of the running machine and the machine in Pause (a decreasing value approaching zero).
4. When the display was about to read 00:00:00 difference*, I put the slave into Play mode and let the synchronizer take control.

This procedure usually produced an initial difference of less than ten frames, but it still took a few seconds for the slave machine to come into frame sync and for its servos to stabilize. This is due largely to the type of sync control we chose to use, i.e. External Playback Sync. This sync input is to keep machines running in a frame sync relationship and so is a very fine control; the speed variation

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*This form is used when referring to the displays of the synchronizer and the video edit controller; both of these display 'minutes':'seconds':'frames'.

possible is quite small. If the initial difference is greater than a few frames, it will take a number of seconds for the synchronizer to get the slave machine into sync with the master. So, a very important rule: when shooting rushes, and at the beginning of a tape, allow a nice long time with time code running for later sync of the material. Thirty seconds is not too long for later peace of mind. This, of course, will depend on the capabilities of your syncrhonizer. If you can tell it to start two machines at specific points of time code, and it will do it all automatically, you have it easy. Systems such as this are, however, rather expensive.

4) It didn't matter whether the PCM or video VCR was the master. I found it to be more convenient to have the PCM acting as master because any major speed fluctuation in the PCM tape caused the processor to mute the signal, and you heard nothing. If the video deck was varying in speed, you got a fluctuating image but could still see what was there.

3. <u>Summary</u>

This system works, once the sync control is set up compatibly with the editor. Playback was accomplished with the sound and image as simultaneous events. Even offsets of a few frames were possible with no visible or audible discontinuity. It seems that we benefit in this instance from a lack of acuity in the watcher. This is understandable, since watching a performance from thirty feet away is equivalent to a one-frame offset in time of sound arrival.

B. TEST #2--VIDEO SHOOT OF A MUSICAL PERFORMANCE

Encouraged by the results of the first test, this system was used to document the Paul Cram jazz group, Kings of Sming, in performance at the Western Front on 21 October 1984.

1. <u>Findings</u>

The system worked well. It was easy to use when shooting and easy to set up for playback. The only aspect to note here is that it doubled the tape cost of the production. We began to realize at this point that a good quality 1/2" VCR would be a good thing to have. This would drop the costs (for the audio) below even that of analog 1/4" tape at 7 1/2 inches per second (19 centimeters per second).

One aspect of the set-up that we discovered during this performance involved bleeding of the sync signal into other sections of the mixing console. The time code was initially run through the mixing board, since the time code was running as an audio signal and this was all normalled into the system. It was found that it bled particularly into the auxiliary mixing busses (being used for house P.A. send) at a low but audible level. Fortunately, it wasn't apparent on the main stereo mixing buss, but this does lead to another important rule: the system should be set up so that the time code signal runs to and from the synchronizer and decks directly. This will eliminate the problems of time code signal getting where it shouldn't.

C. TEST #3--EDITING

1. <u>Introduction</u>

Further encouraged by basic operation of the system so far, it was time to try editing. This was the crucial test that would determine if PCM audio could be used for video production. The lack of a dedicated editing system is the main problem with the E.I.A.J. standard for PCM encoders like the Sony F1 or 701. It is a consumer standard, and therefore perhaps designed not to compete with the professional systems; thus, the only way to edit the digital information is with a standard video editor. The professional system with editor costs over $50 000, which is not in our price range. The main feature of a digital signal editor is that it de-interleaves the data and re-interleaves with the new section being introduced (see section III: INTRODUCTION TO DIGITAL AUDIO). A simple analogy might be that of splicing two pieces of rope--it gets unwound on both sides and then rewound together. With a video editor, we're stuck with tying a knot. The data stream that should be continuous and inter-related is suddenly faced with a contradiction that the processor will validly recognize as an error.

We therefore know we're not doing things the right way. The question is, can we get away with it?

2. <u>Description</u>

For a test signal, the material recorded during the Paul Cram performance was used. The basic procedure was as follows:

1. Both master tapes (video and PCM audio) were striped with video signal and time code to establish a control track for VCR head servos and to allow insert editing. The video signal was colour bars from the Sony DXC-M3 camera, and the PCM signal came from the F1 with no audio input signal (i.e. digital silence).
2. The initial program material was insert-edited onto the video and audio masters.

i. <u>How to determine and set edit points</u>

The time code can only be read by the synchronizer in forward direction at real speed or faster. This means that when shuttling back and forth at low speed, you lose the time code position. This problem was circumvented by using the frame counter on the editor, which will count video frame pulses from an arbitrary zero point. The procedure went like this:

1. Initial Edit

   (a) Determine the edit 'in' point according to the criteria valid; i.e. image or sound.

   (b) Reset the frame counter to 00:00:00 (minutes:seconds: frames) at the edit point and punch it in.

   (c) Put the VCR into play momentarily, then pause. This causes the synchronizer to hold the last time code value in the memory of the display.

   (d) Then subtract the frame counter value from the time code display value.

e.g.	Time code =	5:22:15
	Frame counter =	0:02:10
	Difference =	5:20:05

   Therefore, you know the edit point is set at 5:20:05. This must be done for both rush and master to determine where, by the time code, the edit begins. This must be logged!

   (e) The edit can then be performed.

See also Appendix II--SYNCHRONIZERS, and Appendix III--THE BVE-500A EDITOR.

2. Matching edit

At this point, you've done the first edit (audio or video), and need to do the edit in the other medium. We want to set the edit points to be as close as possible to what was determined in Part 1. The procedure is basically the same.

(a) Run the rush tape to the approximate edit point, watching the time code display.

(b) When very close, put the machine into pause and reset the BVU frame counter to 00:00:00.

(c) Read the time code value and determine the difference between what it is and what it should be.

e.g.	Time code =	5:19:00
	Edit point =	5:20:05
	Difference =	0:01:05 too early.

(d) In search mode, run at slow speed to 00:01:05 by the editor frame counter, reset the frame counter to zero, and set the edit 'in' point.

When this is done for the master as well, you should have an edit being performed at exactly the same time as the other.

3. Check it!

One thing I learned very early was that each edit should be checked as it is done to ensure that sync is maintained between image and sound after the edit point. It is very easy to fix right away, and very complicated later to figure out where you went wrong.

3. <u>Results</u>

It worked. No click, pop, or any other artifact noticeable. However, due to the nature of the music (quite continuous and evolving jazz), it was impossible to make an edit that made musical sense. There was a jarring quality to the edit, and the continuous sound may have been masking any edit point artifacts. It seemed that a more controlled test signal was necessary to really hear what was going on.

D. TEST #4--FLUTE DUET

1. Introduction

This test comprised a special recording session of a flute duet specifically to get material for editing. The musicians performed a piece three times with a different camera shot each time. First was a two-shot, then a close-up on each flautist individually. The music performed was a Telemann flute duet--contrapuntal, overlapping lines, which were quite continuous. Editing procedure was the same as in Test #3.

2. Results

There was no problem making a musically successful edit, though this was of course limited by the differences (of tempo, note releases, etcetera) between the various takes. There was, however, an artifact at the edit points; a small click equivalent to flicking your fingernails together. Not bad, but not that acceptable either. As the edit point went by, the mute light on the processor was lit momentarily.

So, the question was, is this click a part of the muting circuit itself, or is it the result of the muting circuit acting on a signal at that time? Because of the continuous nature of the signal, edits (which were always set at the beginning of a phrase for one part) would always be cutting a sustained note on the other part. What we might be hearing is the turning on and off of that signal, which is a form of amplitude modulation. A sharp cut off or turn on slope could produce an instantaneous wide bandwidth pulse, heard as a click. I assume that the success of the edits on the Paul Cram material (Test #3) was due to the heavy masking potential of the signal (two saxes, bass, drums). The two flutes, on the other hand, were very exposed.

It seemed that I had made a poor choice of test material in that there were no silences in the music at which to try editing. Actually, there was silence at the beginning of the piece where the signal was first put onto the silence recorded earlier, and there was no artifact at this point. Another test was needed; this time with material that had some simultaneous rests.

E. TEST #5--FLUTE AND GUITAR

1. Description

This test was again a special recording for test purposes only, and used the same three camera angle procedure. This time the instrumentation was flute and classical guitar, and the music was more homophonic. What was particularly useful was that it contained coincident breath indications for a slight simultaneous pause. This is where the edits would take place, hopefully surreptitiously. One further positive aspect to the composition chosen was its alternation of lead parts, which led to some natural video edit points.

2. Results

It works. You can actually do an edit in the pauses with no audible artifact. The only problem I ran into here was when I did a succession of video edits without doing the matching audio edit. After I had later performed the audio edits I found the synchronization to be about a second off, and couldn't figure out where I had gone wrong. I knew I had misread the display when logging the edits at some point. Finally I went back and redid those sections, which took less time than I had spent trying to find the mistake.

This brings up one advantage to electronic editing--you never touch the rushes once recorded except to play them. All editing takes place by transfers to the master tapes, so while a major disaster might wipe the master out, you would still have all the original material.

At this point, it was clear that the basic concept of the system--its operation and editing capability--was sound. The next stage was to accomplish a complete production.

VI. VIDEO PRODUCTION--J'AURAIS DIT--GLENN GOULD

A. INTRODUCTION

J'Aurais Dit--Glenn Gould was an audio/video work conceived by Quebec artist Rober Racine and realized as a Western Front Video production while he was Artist-in-Residence in November of 1984. While the tape stands on its own, the theme dealt with the radio documentaries produced by Glenn Gould for the CBC. The techniques and aesthetics of these productions were explored in relation to M. Racine's own work as a visual artist, documentary producer, and pianist. It involved recreations of Gould's unorthodox practice technique, M. Racine's work on his own "page mirroir" project, use of Gould's released recordings, excerpts from interviews with Gould and his technician, John Jessop, excerpts from his radio documentaries, etcetera. It also used some of the sound collage techniques that Glenn Gould pioneered. All in all, a perfect project to undertake as a test of the system.

This is as good a time as any to talk briefly about a few aspects of audio in video. It's neglected. Part of this is due to the very strong nature of visual information tending to subordinate the audio. This production by Rober Racine is in fact the most balanced production I've worked on in terms of the importance of the two aspects. There are times when there is image and no sound, and also times where you have sound but no image. I believe that this balance comes partly from its theme and partly from the artist's experience in music. In fact, this is his first video production.

The main problem I perceive is that the artists working in video have no real conception of what it is possible to do with sound. As a result, they bring fourth generation cassettes or twenty year old discs to be used as materials in their productions. The only solution to this is to somehow expand the availability of audio production facilities and access to the technical staff, and to have greater promotion of education projects.

B. PRODUCTION--AN OVERVIEW

The production procedure can be broken into a few basic sections:

(a) shooting of video material that has no accompanying sound;

(b) shooting synced video/audio material;

(c) audio recording;

(d) preparation of pre-recorded material (editing, equalization) from other sources, such as the excerpts from the Gould interview and radio documentaries, which were dubs from CBC masters.

There is no need to deal with section (a). There was no need even to put time code on these tapes.

The synced video/audio shoots, (b), were all done to PCM audio master with the same procedure described in Test #1, and with the set-up shown in Figure 1. Approximately fifteen seconds lead-in time with running time code was put in advance of any shot action. Because there were multiple takes on each rush tape, the clock was not reset but simply stopped at the end of a take and then started again from that value for the next take.

Production section (c), involving audio only, was a straightforward PCM recording. This time I did put time code on the tape because it was potentially useful later. I'd come to appreciate having the time code available through all rushes just to easily identify the different takes.

The preparation of the pre-recorded material involved editing of the interview material so that salient sections were brought out and condensed for particular effect. All analog material was bounced to a submaster PCM tape before editing into the PCM master. This allowed use of a programmed edit with the video editor/controller.

C. POST-PRODUCTION--AN OVERVIEW

The post-production can also be divided into some basic procedures:

(a) non-synced sound sections using original PCM material;

(b) non-synced audio from analog bounced to PCM digital;

(c) synced PCM audio/video sections (i.e. one audio source);

(d) multiple audio source mixes with a synced aspect; e.g. a shot of M. Racine speaking, with overlays of music and excerpts from interviews.

In situations (a) and (b) it was typical to perform the video edit and then fit the audio to that initial edit point. It was not unusual to let the video run past the point where it was likely to end for that shot, and then insert the next video image into it. As with all edits, it was necessary to carefully log the insert point per the master time code to later perform the matching edit. The analog guide track was laid on the free audio channel (#2) of the PCM master while that edit was made; channel #1 was already occupied by pre-laid time code on both masters.

Situation (c) occurs when you have audio synced to video during the shoot and it is the only audio accompanying the scene on the master. This is equivalent to the sitatuion in Test #5 (Flute and Guitar). An initial edit is performed (usually video), and the time code is used as a reference to make the matching edit (which would be from PCM rush to PCM master). In choosing the edit point, it was, of course, most reliable to perform the audio edit at a point where there was a momentary silence. In this situation, having the analog cue track on the video rushes was necessary for convenient location of possible edit points.

Where you have multiple audio sources that have to be mixed and they include some synced material (situation (d)), the best option is to go to a synced multitrack analog machine. The procedure used in this production was to stripe one track of a Teac model 38 eight-track recorder with time code, and to treat this as a continuous parallel of the PCM master. Thus, you proceed by doing video edits and transfer the audio to the multitrack which is synchronized, or slaved, to the video (see Diagram 3: Bounce From PCM Audio to Multitrack Analog). This is a little more awkward than dealing with video and PCM VCRs in that you no longer have automatic punch-in. You can, though, sync the multitrack to a particular offset between it and the time code of the rush VCR. For example, if you have a particular cue at 10:00:00 (minutes:seconds:frames) of the master, and that audio material comes from a PCM tape at 7:30:00, it is then simply necessary to get the machines running at an offset of 2:30:00 and lay that audio section onto two tracks of the multitrack.

Some awkwardness comes from the fact that with this synchronizer you cannot set a particular offset directly, but must get the machines running and achieve the offset by using two buttons: Increase Offset and Decrease Offset. This points out again the advantage of a goodly pre-action sync period, as it can take a few seconds to accomplish this with this manual method. One advantage here is that the eight-tracks's D.C. motors are much more responsive to control of the synchronizer, and can have a much greater speed variation: ± 50%. The speed variation possible with the BVU VCRs using the External Playback Sync control allows only ± 10%. The limiting factor here is that the VCR circuitry is somewhat 'smart', and if the external sync input signal is beyond certain bounds, the playback servos of the VCR will ignore that signal and use its own internal time base.

Once all tracks have been recorded on the multitrack, it is quite straightforward to get the mix to the PCM master. The first step is to mix to a PCM submaster. (There can be a synchronization problem in this mix, but that will be dealt with in Section E.) All audio tracks are mixed to stereo and routed to the PCM encoder's inputs and thus to the video input of a VCR. The time code track of the multitrack is bounced to an audio channel of the VCR, and then an edit is performed using the video editor to transfer the information from the PCM submaster to the PCM master. No offset is necessary because you have set up the multitrack to be a parallel master and all audio cues are placed in relation to its time code, which is identical to the time code of both audio and video masters. A cue that should happen at 10:05:13 has been recorded at that point on the multitrack.

D. ANALOG CUE TRACKS

All through these procedures, an attempt should be made to keep laying down an analog cue track. This is not always possible, due to some limitations with the BVUs. The simplest situation is when the only signal comes from recording with the shoot. In this case, the original video's cue track recorded during the shoot is transferred to the video master's cue track while doing the edit, and the PCM audio's cue track can be transferred directly from the rush tape to the PCM audio master's cue track while doing the edit. The PCM master's cue track could also be derived from a mix of the PCM material off the rush if there is a problem with the rushes' analog track.

In the situation where you have no original audio with the video shoot, you cannot transfer the guide track. Therefore, you have to use the time code as a reference and do an audio insert from the audio master to the video's cue track under control of the video editor. The only danger here is getting off. With care, any discrepancies between the PCM audio master and the video cue track will be less than two frames, which in these circumstances is negligible.

A good way to check that your PCM master is in perfect sync is to mix it and the cue track from the video master into the monitors. The result of mixing material which is identical but slightly displaced in time is a comb filter effect which very noticeably affects timbre; if the material is displaced far enough in time, you will get an echo effect instead.

The problem with getting a continuous cue track on the video master comes from a crosstalk problem within the BVU's record head while recording. There is a similar audio problem in multitrack technique, usually solved by not transferring material between adjacent tracks. In this case, the problem is that while recording audio on one of the two tracks a large crosstalk component appears on the output of the other track, the one with the time code. This confuses the synchronizer quite terribly. If the synchronizer has control of the slave machine while misreading the data, it will cause the slave to fluctuate in speed. This in turn will cause either fluctuation of the video image or muting of the output of the digital processor. If the synchronizer has the eight-track recorder as slave, it will cause enormous amounts of pitch and time changes. As a result, we cannot record on a VCR audio track while a BVU or the multitrack are required to be running <u>under synchronizer control</u>. In the normal editing situation, the time code is used only as a time/ position reference. The machines are being controlled by the video editor, so this is no problem.

The possibility of crosstalk into the synchronizer's time code send and return lines should always be kept in mind if there is a problem. If the (in this case green) light indicating proper reception of time code into the synchronizer is fluctuating, then something is wrong. This may be miscalibration of levels at some point, or an intermittent connection. It may also be that some other signal is getting into the time code playback input and causing the circuitry to misread the code because the time code is stored on tape as changes of frequency.

A good rule of thumb concerning levels would be to record the time code at a strong level and the analog cue track at a low level. While there doesn't appear to be a problem with crosstalk during playback only, it would be best to play it safe. This is the type of problem that will vary with the video units used. With other VCRs it may be possible to record on one audio channel while using the time code from the other. Something else that would affect the capability of the system would be having a SMPTE (or compatible) system that would use the separate SMPTE track for storage of the time code. If this were the case, there would be no problem with crosstalk between the audio channels. It would also allow the possibility of a stereo audio track on the master and distribution dubs. In distribution this would allow you to have a video tape with stereo analog audio tracks that could stand alone and still have the option of using the separate digital master if the necessary equipment is available to sync a separate PCM audio drive.

E. LAYBACK FROM MULTITRACK TO PCM MASTER

The whole business of using an analog machine in this digital application may seem odd but does make sense. The overwhelming benefit of this whole project is that the audio tracks of the video machines are avoided. The digital technology used enables us to do this, but it does have limitations. Some of these include:

--only being able to do butt edits while going from PCM rush to PCM master;

--only being able to go into record mode on both channels simultaneously.

There are occasions when having the ability to synchronize a multitrack recorder to the video is a pure delight. These include:

--when there are a number of different audio tracks to be mixed;

--when crossfades of different material must be made.

These can be done easily, quickly, and in perfect sync to video while only going through one generation of good quality analog.

Mixing the multitrack tape to a stereo digital submaster did present some problems the first time this was tried. During all the transfers to the eight-track, the recorder was slaved either to the VCR that the material came from or to the video master. The latter usually happened when other analog material was bounced, or if the edit had to be a visual cue. As mentioned before, the eight-track had previously had one channel of time code recorded continuously on it so it could run parallel to the audio or video master through the whole program.

When it came time to mix this material to two-track PCM submaster and then edit this into the PCM master, I discovered that the eight-track no longer run at the same speed. The mix occurred with the eight-track unslaved due to the crosstalk problem mentioned in section D. I knew this should not happen, becuase the time code was also originally recorded with the machine free-running. Somehow, some operating parameter of the multitrack had changed.

I later discovered what this problem was; after the production was finished, in fact. The time code had been recorded while there was no connection to its control input from the synchronizer.

Connecting the synchronizer output to this input, even while the synchronizer was in 'Bypass' mode, caused the machine to run slightly slower. When the transfers to the eight-track had been made, it had been controlled by the synchronizer with a very stable BVU as master. This had prevented any discrepancies when playing the eight-track with the video master.

Obviously, at the time, I had to find some way of having the eight-track slaved to a stable time reference, and this could only be a BVU. One method would be to do the edit directly to the PCM master. The eight-track would be slaved to the master, and therefore no cue track would be recorded. However, this would greatly increase the chance of messing up the next edit, and if the mix didn't go perfectly, it would mean multiple edits at the entry point onto the master. Not desirable. A PCM submaster was definitely the best way. Multiple takes could be performed at no risk to the master, and an analog cue track could be recorded on later transfer to the master.

The method used to produce the submaster involved both VCRs and the eight-track. While programmed to do an auto-edit at the desired entry point, the Player (which had the Video Master on it) was used to control the eight-track. The recorder then recorded the digital signal from the processor on its video track, with the time code from the video master on channel 1 audio, and a cue track on channel 2 audio (see Diagram 4: Mixdown of Multitrack Tape to PCM Submaster Under Synchronizer Control). Since the two VCRs ran in sync and at a very stable speed while under control of the editor, it really made no difference which was master. Any problems would immediately be noticed because the video master was being replayed while the mix was made. It also meant the mix could be made to visual cues rather than by numbers.

F. REPROCESSING PCM MATERIAL

One of the limitations of this PCM recording system is that you cannot alter any of the sonic parameters in digital form. If, for example, a section was recorded at too low a level, there is no way to correct this without going to the analog domain. Here is where knowing your equipment can have an effect. A brief look at the block diagram of the Sony PCM-Fl processor (see Appendix V) shows that encode and decode circuitry are completely separate. It should then be possible to decode PCM encoded material to analog form, process it in amplitude or equalization, and re-encode it to digital. This would require only one PCM processor and two VCRs. The one limitation of this procedure (again referring to the block diagram) is that the meter displays the decode output. To meter the post-decode and processed signal will require the calibration of another meter bridge (preferably Peak Program Meter) to the system.

This technique was used in one section of the Racine work. The section requred a synced sound/video shoot of piano, television, and radio, all at quite a strong volume. So, in my conservative fashion when dealing with digital, I left a lot of headroom to allow for the large peak transients of the material. When this section was later edited into the master audio tape with a direct video edit, it was quieter than the preceding dialogue. The reprocessing technique was then applied by:

--decoding the original PCM recording,

--sending the F1's analog output through the mixer for amplification and metering,

--returning the signal to the F1's analog inputs, and

--taking the digital video output to another VCR for recording.

As this was accomplished, the time code and analog cue tracks were bounced directly between the two VCRs (see Diagram 5: Reprocessing PCM Audio During a Bounce).

Unfortunately, our mixer had only VU meters. This made the calibration of the system more difficult, and the first attempt at this bounce was frequently over-modulated. Fortunately, the one meter on the encode side of the processor is the over-modulation indicator, which flashes red. This, then, will indicate if the signal is too hot.

Once again, as with all sections that needed some handling and couldn't be directly video-edited to the PCM master, a submaster was made. This submaster, once satisfactory, was then edited onto the master in the usual manner. Since there is no generation loss in the material as long as it is recorded in digital form, this is a most satisfactory way to proceed.

Those with sharp wits may already have noticed another possibility inherent in being able to encode and decode material simultaneously; that is, overdubbing other signals with the ancient and honorable technique referred to as 'bouncing'. One could initially record some material in PCM format and then, in the process of a transfer between VCRs, add other new material. In this manner it is possible to do an unlimited number of other overdubs without ever using an analog recorder. The drawbacks of this technique are, however, restricting. Once material has been added to the PCM tape, it can no longer be removed or altered. This requires that the entire finished product be thought out in advance in terms of balance of parts, spatial perspective, etcetera, and that the occasional good performance cannot be used because it is too high or low in the mix.

G. RECORDING AND REPRODUCTION LEVELS

The aspect of relative balances between sections of a work is one that needs some attention in pre-production. One of the hardest habits to break is that of recording all signals at or as close to saturation level as possible. With a system like this, there is actually some useable dynamic range. The question does remain, however, of how much dynamic range (in terms of average volume) is desirable. There will be different factors affecting this decision.

One of the first factors is the kind of playback system envisioned. If replay has to be through a television monitor loudspeaker, you might as well compress and limit the bandwidth. Fortunately, video art is usually distributed in the form of tapes, so there is no difficulty in attaching a decent playback system to the replay VCR. With recent development of the 'hi-fi' Beta and VHS 1/2-inch video decks, it is possible to have a distribution medium that has high quality audio, though there is some compromise in video quality.

Assuming a reasonably high quality reproduction system in an excellent environment, there are still problems for the audio engineer. At least they are no longer to do with the especially narrow noise/distortion window of audio in video. In other words, we don't have to worry about low level signals disappearing into the tape noise while high level signals are terribly distorted (at least not on the digital tape). We now have to consider the following questions:

--how loud will the typical listener want the audio?

--how much amplifier power will that need for a typical hi-fi loudspeaker?

--how quiet should one assume the typical background noise level to be?, and

--what typical amplifier power should be assumed?

From the one subjective parameter we get a number of physical parameters to consider.

We need a reference from which to work, so we'll start with amplifier power. We'll assume a one hundred watt RMS amplifier with 3 dB peak headroom. That means the amplifier can handle two hundred watt surges on peaks of the program. This would class as quite a good amplifier. If we allow the universal 10 dB for typical program peaks, it means that the amplifier can coast at 20 dB RMS for our loudest average level. If there was a 40 dB ratio between our loudest average program level and our quietest (this represents a power ratio of 10 000), the amplifier will put out 0.002 watts RMS at its lowest. I suggest that with a typical loudspeaker in a typical ambient noise level it would be impossible to hear the quietest level. So, there are definite physical and physiological limits as to how much useable dynamic range there is.

In relation to the F1's operation, I suggest at least 15 (or even 20) dB for peak headroom. Our loudest average level would now sit at -15 dB. If we considered 10 dB an adequate ratio between loud and average, we would then record dialogue, for instance, at about a -25 dB average level. And on it goes--there are definite problems in metering at low levels (a quiet signal will of course be below this). It is hard, I can attest, to break years of conditioning and start recording signals at these levels.

So far the question of dynamic range has been entirely physical, not subjective. Listener preference (in terms of average levels) is a rather vague area. For example, if there is a shot of a jack hammer, do you want to be able to reproduce that as a real level in playback? This is rather unlikely; rather, you would set the playback level so it seemed loud. If you then had a whisper recorded at the proper intensity relationship to the jack hammer, it would be inaudible. So there has to be some compression of the dynamics; all audio engineers will admit to this 'cheating'. If you record a whisper, it is usual to close-mike it and let the quality of the voices carry the effect. The listener suspends disbelief while we play with perspective.

The real question is: "How does one work with dynamic range in production to allow as much change above and below the average as is desired psycho-acoustically?" The only suggestion I can make is to calibrate your ears to the recording/reproduction system. This sounds a little odd, but it is possible. The first step would be to record on the F1 a nice, even, musical signal at the -15 or -20 dB level. Then, play this back to establish the loudest average playback level. Just turn it up until it seems good and loud, and then tape down or mark the monitor level control. From then, work by ear. It's unlikely that you will want dialogue at that loudness level, so naturally you'll record it at a lower level. Monitoring should always be through the PCM processor when possible, to avoid getting differently-set gain stages upsetting your reference.

With this procedure you should be able to avoid the problem I experienced of recording dialogue too hot and then finding that I didn't have enough dynamic range left above when I wanted something loud. With low level signals, it will be natural to fit them into your own hearing preference, and perhaps to equalize them to compensate for our hearing behaviour at low intensities, rather than to turn up the monitors.

Dealing with high dynamic (and therefore recording) levels, and allowing the headroom to achieve them in a production, is even more important in digital recording than with analog. With the digital encoding process we have a fixed, immutable, maximum level that can be recorded. This is due to the fact that the analog-to-digital converters are referenced to a certain voltage input to achieve full modulation; that is, the use of the highest number they have available to use to represent the amplitude of the input signal. Once this limit is past, the onset of distortion is extremely rapid and very noticeable because the artifacts it adds to the signal are quite non-harmonic. With analog recording we could, if necessary, adjust the recording flux to achieve a higher level on tape for a particular section and accept the resultant, basically harmonic, types of distortion that might occur. With the digital format, we have such a nice amount of dynamic range available to use with no perceptible noise that the best procedure is to allow the headroom above the standard operating level that might be necessary to accomodate the loudest desirable signal level and still have the necessary peak headroom for transients.

The final problem is playback. Now that you are using the dynamic range, how do you give a playback intensity level indication to those showing the tape at a later time? If, for example, the tape begins with a very quiet or loud level, how are they to know it should be quiet or loud, and not medium? A sine tone is fine for calibration of the processor outputs for the best interface to other following equipment, but gives no useful sound level indication. The loudness of a pure tone, especially 1 kHz, can be hard to judge. It should be recognized as the reference level and therefore the highest likely signal level. However, there could be different intents for a signal recorded at this level. The entire soundtrack could consist of whispers meant to be barely perceptible. The tradition is to record all material on tape as strongly as possible and to adjust the final volume level setting on playback. This is the only way to deal with the limitations of analog recording tape. There is now the option of reproducing very quiet signals with no background noise (other than that recorded) and still having enough dynamic range to present quite an impressive explosion (within the limits of the reproducing system).

The only suggestion I can make right now would be to put some actual reference material on the tape along with directions. The reference could be a verbal count, or some music, with a description of its relative loudness. Alternately, directions could be given about a section of the tape; e.g. "Amplifier should be set so initial dialogue is a clear conversational level. The loudest level is at ten minutes and thirty seconds." This would also be a useful indication for a tape produced conventionally. Actually, I would be quite happy to get any reference tone on the distribution copies of tapes I am responsible for showing. This should definitely be a standard practice.

H. DETAILED DESCRIPTION OF THE PRODUCTION

1. Scene 1

This scene consists of one continuous video shot with one continuous section of music. The camera starts on a long shot of a piano with a radio and telephone on it, zooms in, and pans right. The beginning cue for audio was immediately after the shot begins, and the end cue for the scene is when the pan has taken the radio completely out of the shot. This happens shortly after the musical excerpt ends. The total length of the scene was determined by the duration of the audio excerpt. Visual and audio cue convergence was to have the music end just prior to the radio leaving the shot. Procedure was as follows:

(a) Duration of the musical excerpt was determined to be about 3'20";

(b) Video rush entry point was set 3'20" before the end video cue (radio pans out);

(c) Master entry point was set at 2' by the time code.

(d) The video edit was performed leaving enough material after the end cue point to allow the next scene to be inserted into it.

(e) The PCM submaster containing the excerpt was placed on the Player VCR (this had been bounced from disc earlier). The PCM audio master was placed in the Recorder VCR.

(f) The PCM master edit start 'in' point was set using the time code as a reference; i.e. 02:00:00 time code (2 minutes, 0 seconds, 0 frames).

(g) The rush (PCM submaster) edit 'in' point was set using the analog cue track as a guide. An initial section of about twenty frames of silence was allowed before the music entry in case of muting of the track later.

(h) The edit was performed, again allowing the signal to run past the point where the next edit would be performed.

And finally,

(i) audio and video masters were set up and run under synchronizer control to check the scene. This was done after every appropriate edit.

2. Scene 2

This scene involves an interview of Glenn Gould by the CBC concerning the radio documentary, Strauss: The Bourgeois-Hero.

This interview excerpt section is approximately 1:47:20 in length, and runs straight through. The interview was animated by the use of cuts between still images of Gould and Strauss. In this case the initial starting cue was the end of the final image at the end of Scene 1 (the radio panning out of the shot). Image I of Scene 2 was a close-up shot of a radio tuner as the introduction of Mr. Gould is heard, thereby conveying that this is a radio interview. The remainder of the cuts are to audio cues. For example, image II (a still of Gould in a radio studio) comes in with Gould's first words. The remainder of the images are stills of Strauss, used to highlight portions of Gould's comments on him, mixed with another image of Gould. These are images III, IV, and V. Then the interview turns to discussions of Gould's previous documentaries, with an image of Gould working in a radio studio, and finally switches to a shot of M. Racine's own "Page Mirroir" project while the discussion centers on the spoken word counterpoint technique which Mr. Gould used in a previous production, The Idea of North. This scene ends with the "Page Mirroir" gradually going out of focus as the sound level fades. The end of the audio fade is then the cue to the beginning of Scene 3. Achieving the coincidence of audio and video at this point is a good example of how it is often necessary to determine edit entry points on the Player (rush material) and Recorder (master).

In this case it was desired that the focus dissolve would begin on the phrase ". . .exemplary clarity. . ." at about 04:46:00 time code on the master. This phrase was also used as the cue to begin a slow fade of the audio when bouncing from the analog half-track tape of the interview to the PCM submaster. This was then edited onto the PCM audio master and an analog cue track was edited onto the video master. This was subsequently used to determine the video edit points for this scene. The video rush material consisted of a lengthy still shot of the page (a couple of minutes) which then was slowly unfocused (about six seconds) and left for a further one minute. Therefore, the video cue to work against was within the section we wanted to insert.

To achieve the desired relation between audio and video, we did the following:

(a) determined where the audio cue was on the master--04:46:00 time code;

(b) determined the desired entry point of the image on the master--04:13:03 time code--and reset the Recorder frame counter to 00:00:00 at this point and entered it as the Recorder 'in' point;

(c) determined the length of image necessary before the cue--

i.e.

$$\begin{array}{r} 04{:}46{:}00 \\ -\ \underline{04{:}13{:}03} \\ =\ 00{:}32{:}28; \end{array}$$

(d) found the visual cue on the video rush, and reset the editor frame counter to 00:00:00 at that point;

(e) rewound the video rush by the amount determined in (c), i.e. to -00:32:28;

(f) entered this as the Player 'in' point;

(g) performed the edit.

Note: no 'out' point was necessary at this time. This can be done manually while the edit is being performed. It is simply necessary to allow the image to run past the point where the beginning of the next image will be inserted.

3. Scene 3

Scene 3 was quite straightforward, consisting of a single, non-synced audio source against just two video images. The audio portion was an excerpt from the radio documentary, The Quiet in the Land. This had been bounced from half-track analog to PCM sub-master and then edited onto the PCM audio master. The video consisted of a slow pan of houses in North Vancouver. One edit was performed because there was not enough desired material in the rushes for a single continuous shot.

The audio was done first, since this determined the length of the scene. The 'in' cue was a gate squeaking while opening, and the 'out' cue was a car starting.

4. Scene 4

This is the first section of the production that used sound from more than one source, and therefore required the use of the synced analog multitrack or the bouncing technique. In this production the synced multitrack was used exclusively because of its convenience of operation and the fact that the additional material aside from the sound recorded with the shot came from non-digital sources.

One of the very nice features of the multitrack is that it can be synced to the video master and all sound/video correspondences are immediately apparent. The first step taken was to bounce the PCM audio master that existed so far to two tracks of the multitrack. Though this wouldn't be used later, it enabled us to audit the complete production as editing proceeded, without resetting the synchronizer to operate the video machine. There were, of course, two different interfaces provided with the synchronizer one that was frequency variable for the VCR, and one with a variable D.C. output voltage to vary the drive speed of the multitrack. As has been previously said, the multitrack tape was striped with a continuous time code so it could run parallel to the video through the whole production if necessary.

The three sources of audio for this section were:

(1) One channel of sound recorded during the video shoot. This was of M. Racine cutting words out of a music dictionary with an X-acto knife. It was recorded with a PZM microphone placed on the table immediately beside him but out of the shot.

(2) Two channels (stereo) of an excerpt of the Beethoven Sonata for Piano, opus 27, no. 2, performed by Glenn Gould (C.B.S. M37381). This was on a cassette provided by the artist.

(3) One channel of an interview by M. Racine of John Jessop, who was Mr. Gould's CBC technician for the radio documentaries. This came from a half-track analog tape recorded by Mr. Jessop in Montreal, and was also provided by M. Racine.

Procedure was as follows:

(a) The desired section was video edited onto the video master.

(b) Audio was bounced from sync sound PCM rush to eight-track at the same time code values at which the video edit was performed. The multitrack was slaved to the PCM rush during this transfer. This required use of the time code offset capability.

(c) The multitrack was then set up as slave to the video master to check synchronization.

(d) The music and interview sections were transferred to the eight-track as image and multitrack were running in sync. This involved cueing up the material, setting levels, and manually starting the machine with the original sound at the correct time. If this didn't work out to taste the first time, it was reset and recorded again.

(e) Once the correct time relationships of the various tracks had been achieved, they were mixed to a stereo PCM sub-master, as has been detailed in section E.

(f) The submaster was edited onto the PCM audio master. The edit from Scene 3 to Scene 4 is very tight and shows what can really be done with the system. The edit actually cuts off the trailing end of the envelope of the preceding sound. This was being quickly faded in the transfer to PCM submaster from analog original, hoping to get to silence before the edit 'out' point. The entering audio is of very low level background noise. If the tape is scanned at very slow speed, it can be determined that there is a two frame silence (about sixty milliseconds) between the two scenes. This is imperceptible, and the two scenes seem to be butt edited with a conventional splice. Actually, a splice might be audible at this point; even analog can have limitations. The 'out' point audio edit of this scene is also worth noticing, but will be dealt with in the next section.

5. Scene 5

Scene 5 presents some interesting problems of description. This is due to the rather unusual structure of the relationship between audio and video material that M. Racine wanted. Rather than the conventional arrangement of audio and video beginning and ending together, they are terraced.

The first shot and audio are of M. Racine discussing some of the techniques used by Mr. Gould in the documentaries. Over top of this is an excerpt from the "Trio" section of the documentary The Idea of North, which uses Gould's vocal counterpoint technique. The next image is of a transcript of a discussion between Gould and John Jessop, while M. Racine's voice continues. Then the audio changes to another excerpt of M. Racine interviewing John Jessop. While this audio continues, the image changes to another shot of M. Racine working on the "Page Mirroir". This shot continues while the audio changes to an excerpt of Strauss: The Bourgeois-Hero. Finally, again with the "Page Mirroir" shot, we get a vocal counterpoint of Kate Craig and Rober Racine reading Gould's description of his practice technique when encountering a block. This is actually very fugal, with M. Racine starting the reading in French and then Ms. Craig coming in reading the same text, but in English. This forms the introduction to the next scene, wherein M. Racine re-enacts the practice technique described.

There are two sections of sync sound in Scene 5. The first is M. Racine speaking at the beginning of the scene, and the second is M. Racine working on the "Page Mirroir" at the end of the scene--11:37:18 to 15:47:11. These two sections required use of the time code and synchronizer in the transfer from rush to eight-track tape. As in other synced sound sections, it was normal to perform the video edit first, log the time code values for rush and master, and then duplicate the edit for the audio--in this case treating the eight-track as master. The only difference when recording synced audio to the eight-track is that it is slaved to the rush tape with time code offset as described in section V, part E: Layback From Multitrack to PCM Master.

The non-synced audio was bounced to the eight-track while it was slaved to the video master. The non-synced sources containing the Idea of North excerpt, the Jessop/Racine interview, and the excerpt from Strauss: The Bourgeois-Hero were cued and started manually on the half-track recorder with manual punch in and out on the eight-track recorder. Material was recorded at full level and mixed appropriately with fade ins, fade jouts, crossfades, panning, etcetera, on transfer to the stereo PCM submaster.

6. Scene 6--Piano Experience

Here we again return to a synced audio/video shoot with original audio done as a PCM recording and no other audio sources mixed in during post-production. So, as far as editing is concerned, it is very straightforward. In this case the video was put on the master first, using the analog cue track to ensure that the edit point is before the audio begins. The audio was then edited onto the PCM master to matching time code values.

This scene is the re-enactment of Gould's practice technique, described in the fugal reading at the end of Scene 5. The shot is of M. Racine sitting at the piano with his back to the camera, with a television to his left and a radio to his right. As he plays, he turns on first the radio and then the television. The whole scene is really a performance; the desired volume levels of the television and radio were marked and then adjusted to these levels during the shot. The miking consisted of a spaced stereo pair of AKG-414 condenser microphones in omni-directional mode, one on either side of the piano. This produced a natural panning of the three sound sources due to amplitude and time of arrival differences. The television was left, the radio was right, and the piano centered and fairly wide. A little more distortion on the radio would have been more interesting, but it was out of balance at the level required for that.

One of the funny aspects of this shot is that if you look closely at M. Racine, it looks out of sync. However, you have my assurance that all audio in this shot was done live, and you can actually see that the sync correspondence is correct from the television.

The only other aspect about the editing of this scene is that it required reprocessing during a digital-to-digital bounce. The reason for this was that it was recorded more quietly than the dialogue sections preceding and following it, whereas it was meant to seem loud in comparison. The technique to accomplish this has been previously discussed in section V, Part F: Reprocessing PCM Material (see also Diagram 5).

7. Scene 7

Scene 7 consists of a shot of M. Racine again at the table with the "Page Mirroir". This time, however, there is no synced audio at all accompanying the shot. The camera begins a gradual zoom in, and part way through this is the visual cue to begin another excerpt of the Jessop/Racine interview. In this case, the half-track master of the interview was edited to assemble the desired statements by Mr. Jessop with no comments by M. Racine. This was bounced to a PCM submaster and inserted onto the PCM audio master.

8. Scene 8

Scene 8 is one that required a fairly detailed integration of pre- and post-production planning. The concept was to have a series of dynamically changing video mixes of "Page Mirroir" and the score for Bach's Goldberg Variation #6. This was to be done so that there was a visual gesture for each phrase of the music. In order to accomplish this, it was necessary to do the video mixes to the music for each phrase individually (as many takes as necessary), and then edit the image to a complete copy of the music on the analog cue track of the video master. Finally, the music was also inserted onto the PCM audio master to match the analog cue track on video. In retrospect, it would have been better to do the PCM audio master first and then, using the time code as a guide, lay down the analog cue track as an auto edit from the PCM audio to video cue, and then proceed with video editing. With the procedure that was used, it required a few attempts and checks before the exact correspondence necessary was achieved.

During the initial shooting M. Racine, who was operating the video special effects generator, also had the music on cassette beside him so that he could conveniently rehearse the effect with each phrase. When the video was recorded, this cassette was transferred to the cue track on the video rush tape. This later served as a guide to getting the video for each phrase in sync with the complete music already on the video master cue track.

9. Scene 9

Scene 9 once again involves the "Page Mirroir". This time it is a close-up of one page while M. Racine fills in certain letters with gold ink. There is no audio for the initial section. The cue for the first sound is the pen reaching a certain letter. The audio for this section, plus the following titles and a section of no image (black) after the titles, is a composition written and performed by M. Racine, entitled Lieder. This was recorded at the Western Front for this production. The recording was direct to digital format stereo from a mix of three microphones. These were two spaced AKG-414 condenser microphones in omni-directional pattern and one PZM placed on the floor beneath the piano. Two takes of the composition were recorded, the second being used with no editing required.

This music plays through the last shot and the titles, after which the image goes to black. The music continues with no image until the end of the composition, with the completion of the music being the end of the production.

VII. CONCLUSIONS

It has been shown that it is indeed quite feasible to have a parallel PCM digital audio system functioning as the master audio medium for a video production. This system, comprised of an E.I.A.J. standard PCM processor, a time code-based synchronizer, and an automatic video editing system, can fulfill the necessary functions of:

--recording PCM audio in sync with a simultaneous video recording;

--editing of PCM material in its video-encoded form without a dedicated edit processor;

--editing of PCM audio and video at different times using a recorded time code as a guide to make edits which have the desired time relationships;

--replaying edited PCM audio material in synchronization with edited video material.

An issue which must be addressed is the extent to which the findings and procedures described here are transferrable to other facilities and equipment. This is best covered by a recapitulation of the equipment used, its capabilities, and possible substitutes.

The Sony PCM-F1 digital audio processor is built to a standard which is shared by a number of devices made by other manufacturers. In fact, it appears that many of these other devices are either a repackaged PCM-F1 or employ the integrated circuits engineered by Sony for the F1. Material processed by the PCM-F1 is therefore playable on these other processors. According to the documentation on the PCM-Fl processor found in Appendix V, use of the 16-bit resolution mode available on the PCM-Fl does not affect the ability of other processors to replay the material. The digitized signal will of course have reduced error correction capability, due to the sacrifice of error correction code to provide for the enlarged data words (see Appendix V).

While the format of the encoded data output to a video recorder and the amount and type of error detection and correction is specified by the standard involved, there is still room for the actual circuitry used to implement these functions to be proprietary. The efficiency of the error correction, the accuracy of the analog-to-digital and digital-to-analog converters, and possible inclusion of other error concealment methods may result in differing capabilities among the various processors. Since the editing process is disturbing the data format quite severely, it should not be assumed that another processor will function in the same manner as the PCM-Fl. Allowance should be made for the possibility of both better and worse performance in this respect when using processors made to conformity with the E.I.A.J. standard.

These E.I.A.J. standard processors are at present the lowest-cost devices to provide digital audio recording capability. There is, however, at least one professional system which would provide identical function with greater fidelity and with a dedicated editor. This system consists of the Sony PCM-1610 processor and DAE-1100 digital audio editor. As with the PCM-Fl, data are formatted into a video signal for storage on a video tape recorder. Another possibility in a professional quality digital recorder would be a

DASH (Digital Audio Stationary Head) format recorder, such as the Mitsubishi X-80A. This also has provision for a time code track to enable synchronization to another recorder, and which could be used as an editing guide. In this case, editing takes place as a conventional tape cutting and splicing operation. Those interested in the possibilities of these two recorders should contact the respective manufacturers. It should be noted that replacement of the PCM-F1 with either of these professional devices involves a major increase in cost.

The Omni Q TL1 and TL2 modules used in this project are, again, low-cost devices. Neither the time code data nor the output modulation process are to the SMPTE standard. It also does not have the circuitry necessary to read the time code at slow search speeds, nor can it remotely control the drive functions of a video recorder or, except in a very rudimentary manner, an analog audio recorder. It is the method of overcoming these problems which forms a major portion of the operating procedures described. I would therefore recommend that this is the area where the additional cost of going to a more sophisticated SMPTE standard synchronizer is well outweighed by the ease of operation which would result. An additional benefit would be the transferability of tapes to other video and audio facilities, as the SMPTE standard is the accepted industry standard in North America.

It should be noted that in this project there was no attempt at an initial video frame sync between the PCM audio and the video image signals when originally recorded. This procedure was undertaken in the project described in the article "Getting Started in Digital Recording and Post-Production", by Bob Katz, found in the bibliography. It was not attempted here because there was no absolute need found for it in the initial tests, though it would certainly not be a disadvantage.

So far the only type of time code considered has been longitudinal time code. This type of time code is recorded on a continuous analog track. There is another, more recent, SMPTE time code standard for use with video that is referred to as Vertical Interval Time Code, or VITC. This type of time code is used exclusively on video recorders, as it is modulated directly onto a portion of the video waveform for each field. This offers many advantages when dealing with video images. For example, the tape may be in Pause mode, but the rotating video heads which continue to scan the video information can still read the time code. Also, while the position of a frame cannot be identified with longitudinal time code until that frame has gone by and the complete time code word has been read, with VITC the code is on the first part of the video waveform. I have not investigated the aspect of using Vertical Interval Time Code with video-formatted PCM digital audio. This is mainly due to the fact that since it involves modulating the code onto the video waveform itself, it must contain high-quality video circuitry and is therefore still quite expensive. I am, however, somewhat skeptical that unless the decoding circuitry of a digital audio processor is designed to take a signal of this type into account, the use of it would likely be detected as a gross and continuous error in the signal, and the processor would go into a permanent mute. Of course, with the DASH-format recorders the use of VITC would not be possible, since it is not using a video waveform. This also applies to standard analog audio recorders.

The video cassette recorders used in this project were broadcast-quality 3/4-inch recorders--the Sony BVU-200B model. These machines are now somewhat dated, but are still capable of high-quality video performance. These recorders were used in conjunction with the Sony BVE-500A Automatic Editing Control Unit. There is no problem in using a 1/2-inch consumer format video cassette recorder (Beta or VHS) to record the digitized audio output of the PCM-F1. I have recorded hundreds of hours of material with these formats and have encountered no problems. The need for more sophisticated video recorders comes with the desire to edit the NTSC video signal of the digitized audio.

In this area we are again looking mainly at increasing equipment costs to get added features and more up-to-date technology. Even with the 1/2-inch formats it is necessary to go to the industrial models to get the ability to operate the video recorders under control of an edit controller. Here we would be looking at systems such as the Sony SLO-383 Betamax video cassette recorder combined with the Sony RM-440 Automatic Editing Control Unit, or the Sony GCS-50 SuperBeta professional series recorder (which has Beta HiFi capability) and the Sony RM-E50 editor. In the VHS format, a possible system could be comprised of the JVC BR-8600U videocassette editing recorders with the JVC RM-86U Editing Control Unit. All of these systems have framing servos to ensure correct odd and even field sequence is maintained at the edit points, and flying erase heads to enable insert as well as assemble edits.

One aspect of the 1/2-inch formats which must be noted is that their edit accuracy is not typically as great as the 3/4-inch format. For the 1/2-inch systems mentioned above, the edit accuracy is plus or minus three frames, which is approximately one hundred milliseconds.

An editing system based on either of thel/2-inch formats would of course be of lesser cost than the 3/4-inch U-matic systems, and quite considerably less expensive than the one-inch tape size systems. It will be necessary, however, to consider and investigate the individual 1/2-inch systems available in regard to the operating features necessary for this application. The primary considerations would be the capability of the system to perform insert edits and the capability of the recorders to be controlled by an external synchronizer. Other factors would include the availability of a separate time code track (so one of the audio channels does not have to be used), or if there is not a time code track, what effects the level of interchannel audio crosstalk has on the time code signal and the assembly of the analog guide track necessary for editing. See Section VI, Parts D and E, for reference to these factors in the video production described. If one were interested in an audio-only application, the main consideration would be the insert editing capability. Another important consideration, no matter what format of video recorder is used, is whether it is possible to disable any video dropout compensation that might be included, as this type of signal correction (usually simply substituting the previous video line) would only cause the processor to see an even larger error than the original dropout. This should also be considered if using a video editing system that has a time base corrector as a normalled piece of equipment. These frequently employ a circuit of this type, and there is some question as to whether their normal function of stripping and regenerating the video sync pulses is also incompatible with the video-formatted digital audio signal.

One further aspect of this production, and one that has not been dealt with very much in this paper, is the studio facility itself. The arrangement of the patching and switching systems for routing audio and video signals will make a big difference in the ease and efficiency of operations. Some of the necessities include a good pair of loudspeakers with sufficient amplification to power them, and an audio switching system to the monitors which permits easy monitoring of both the Player's and Recorder's analog guide tracks, as well as the output of the PCM processor. Also, arrangements must be made for the type of signal routing necessary to the assemblage of the guide track on the master PCM edit tape. All of these requirements make a mixer with an adequate monitor stage and signal routing capability a practical necessity. The video system interconnection is less demanding due to a lower number of interconnection possibilities. However, addition of a video signal patching system would make the process of changing these interconnections much easier and more reliable when necessary.

To close this paper, I would like to add two brief statements. The first is that if you have an idea to be realized, the best procedure is to go ahead and do it and not listen to all those that say it can't be done unti you find out for yourself. The second is that in today's economy, with its rapid technical advancements, the emphasis for a manufacturer is often on an application that serves the greatest number of possible customers. Unfortunately, this may neglect those of us in areas of production and other creative media in the supply of specific cost-effective products. It does, however, often provide good tools if the available equipment can be adapted to our needs. This report details such an application involving the Sony PCM-F1, but there are surely more such opportunities available now, and more coming.

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APPENDIX I

EQUIPMENT INTERCONNECTION AND INTERACTION

When engaged in production of video, it is natural to organize an equipment set-up and interconnection which (ideally) will work for all situations.

The most common of these situations are:

--recording synced PCM audio and video with time code;

--playing back the above under synchronizer control;

--recording PCM audio alone;

--bouncing to and from PCM and analog audio recorders;

--editing PCM audio and video;

--reprocessing PCM audio by decoding and re-encoding during a bounce between VCRs;

--transferring analog cue tracks from rush to master;

--transferring time code; and

--monitoring all audio and video signals.

It was impossible to satisfy all the required situations with one equipment interconnection in our facility. A comprehensive, understandable audio signal patch bay made changes in audio routing easy. However, changes in video signal routing had to be made by changing cables. This was not too difficult because of the lower number of interconnections possible. However, the required changes on the video decks, such as dropout compensation, line input or dub input, external playback sync, and the movement in signal lines, always had the opportunity for error. Aside from simply making a mistake, there is also the problem that the video recorders' servo systems can be disturbed by inputs not selected or in use. This could happen, for example, by having a video line input to a deck while attempting playback. The line input (from, say, the PCM processor) can disturb the capstan servo (see BVU-200 manual, pp. 1-16).

This type of problem can be solved by following one simple rule: if not using a video input on a video recorder, don't have anything connected to it.

By following this rule when changing between different equipment interconnections and always checking the status of other selectable operating parameters (particularly the tracking and dropout compensation), you won't have any problems.

Note that these are problems specific to the video systems. The audio channels for time code to and from the Omni Q synchronizer and audio sends and returns from the analog cue channel to the mixer may be left in place.

Other variables which may cause problems if not checked with each change in set-up are:

--Tracking Adjustment: if this has been moved to enable playback of PCM material and not reset before recording (PCM or video) again. This is important! If you end up with a series of edits that each require a different tracking setting, you will definitely have problems.

--Drop Out Compensation: if this has been turned off, as it should be when working with PCM material. If the drop out compensation is left off when once again doing image edits, the video signal will be much noisier and tape drop out much more noticeable than it has to be.

Description of Diagrams

All diagrams show video monitors connected to both VCRs at all times. This is easy with our set-up, due to having two video outputs available on the Sony BVUs. With this arrangement you have instant image monitoring for video edits, and the ability to see the quality of the PCM audio signal.

Diagram 1. Recording & Reproducing Synchronized Audio and Video

(a) Microphone signals are routed to the mixer and mixed to two channels.

(b) This two-channel signal is routed to the PCM processor for conversion to a video-encoded digital audio signal; from there, the video is sent to the video line input of a VCR which, in this case, is the Player of the editing pair. The video output is routed to the decode side of the PCM processor and from there the signal (now analog audio) goes to the mixer for monitoring. When recording, one should always monitor the output of the PCM processor. This also means the set-up is instantly ready for replay.

(c) A mono mix of the microphone signals is sent to channel 2 of both VCRs to act as an analog guide track for editing later.

(d) The time code generated by the synchronizer is recorded on channel 1 of both VCRs. The diagram shows the return from channel 1 of both decks to the synchronizer, so you are instantly ready for synchronized playback.

(e) The camera signal is routed to the video line input of the other VCR, the Recorder of the editing pair.

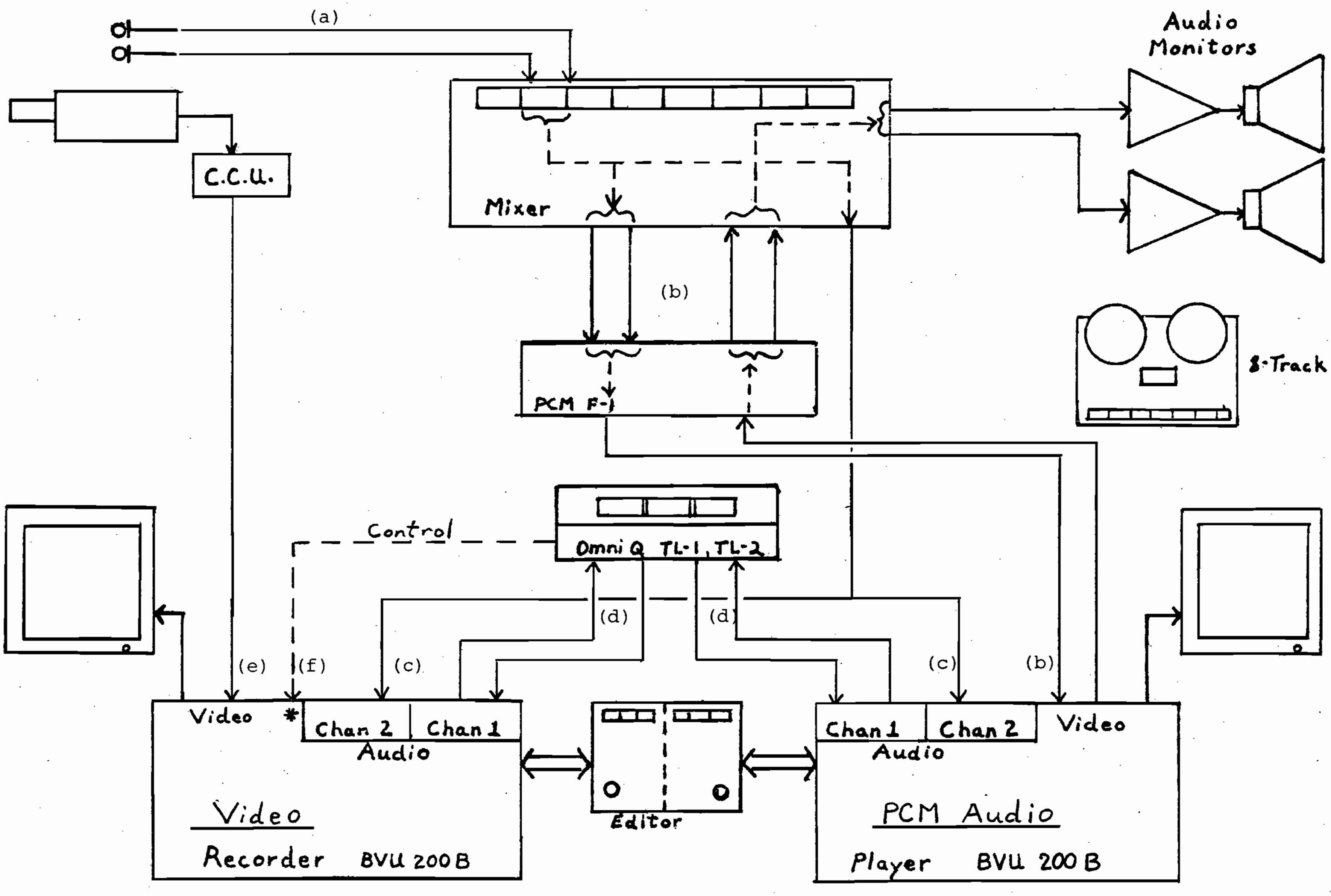


Diagram 1. Recording / Synchronized Audio and Video / Reproduction
<u>During Replay</u> the synchronizer control line is connected to the External Playback Sync input * of the slave.

(f) When replaying the recording, it is necessary to connect the control line from the synchronizer to the control input (External Playback Sync) of the 'slave' VCR. The 'slave' should always be the Recorder of the editing pair (see section V-A-2, part (2)). Also, when replaying, the video input lines should be disconnected to avoid the possibility of capstan drive disturbances. The outputs from both #1 audio channels can be left connected to the synchronizer at all times.

(g) Settings:

Drop Out Compensation:	'ON' on video deck 'OFF' on PCM deck
Tracking:	Reset on both decks while recording; adjust as necessary for PCM playback
Video Input Select:	'LINE IN' on both decks
Playback Sync:	'EXTERNAL' on Recorder BVU when playing back under sync control; 'VIDEO' on PCM Audio VCR (Player BVU). When recording, this should have no effect, but the synchronizer control line should be disconnected!

Diagram 2. Video Image Editing & PCM Audio Editing

(a) Channel 2 (the analog cue track) of both VCRs are going to the mixer for easy monitoring and location of edit points. With this arrangement and the appropriate mixer, it is then very easy to route the output of the Player's cue track to be recorded on the Recorder's cue track. The Recorder will have the Master Edit on it--audio or video. This way, both cue track and master edits can be performed simultaneously.

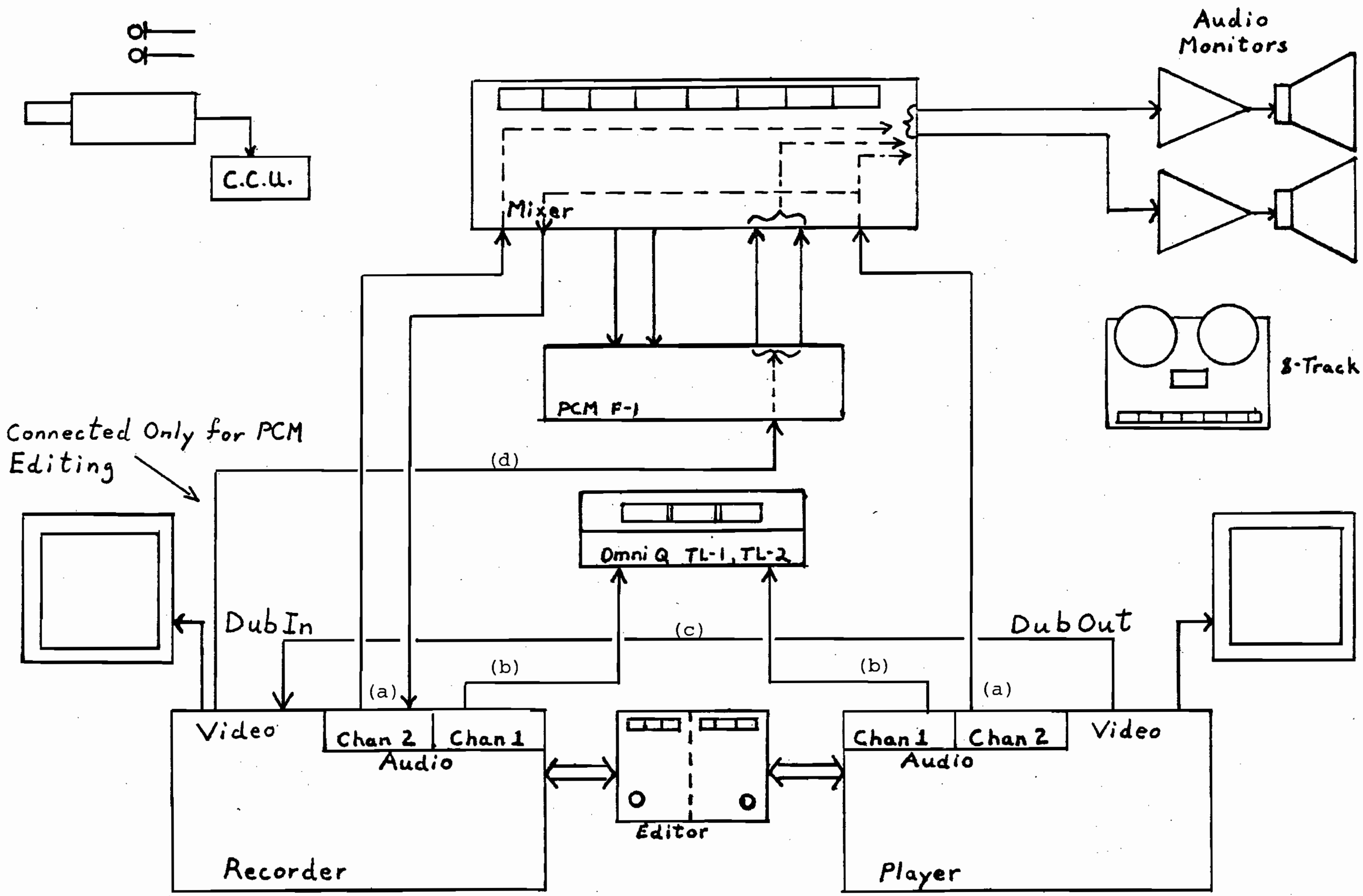


Diagram 2 . Video Image Editing. PCM Audio Editing.

(b) The output of channel 1 of both VCRs goes to a synchronizer input for display of time code. This display is used (in conjunction with the frame counter of the editor) to determine where the edits are occurring (see section V-C-2).

(c) Transfer of the video signal (image or PCM audio) takes place along the special Dub cable from the Dub Out of the Player to the Dub In of the Recorder. This allows much higher edit quality than the straight Line Out to Line In arrangement.

(d) When editing PCM audio, the PCM processor is on the video line output of the Recorder VCR. This way, one can preview the edit to some extent and instantly review the edit.

(e) Settings:

Video Input Select:	'DUB INPUT' on Recorder
Drop Out Compensation:	'OFF' on both decks for PCM editing
	'ON' on both decks for video editing
Tracking--Player:	Adjust as necessary for PCM playback
	Should be 'reset' for image playback
Tracking--Recorder	Should always be 'reset' on Recorder <u>during edit</u>!
	Adjust as necessary for replay to check edit

<u>Diagram 3. Bounce from PCM Audio to Multitrack Analog</u>

(a) Channels 1 to 7 are routed to and from the mixer and multitrack through the dbx noise reduction system. This way, the decoded PCM audio can be routed to any channel of the 8-track while the other channels can be monitored.

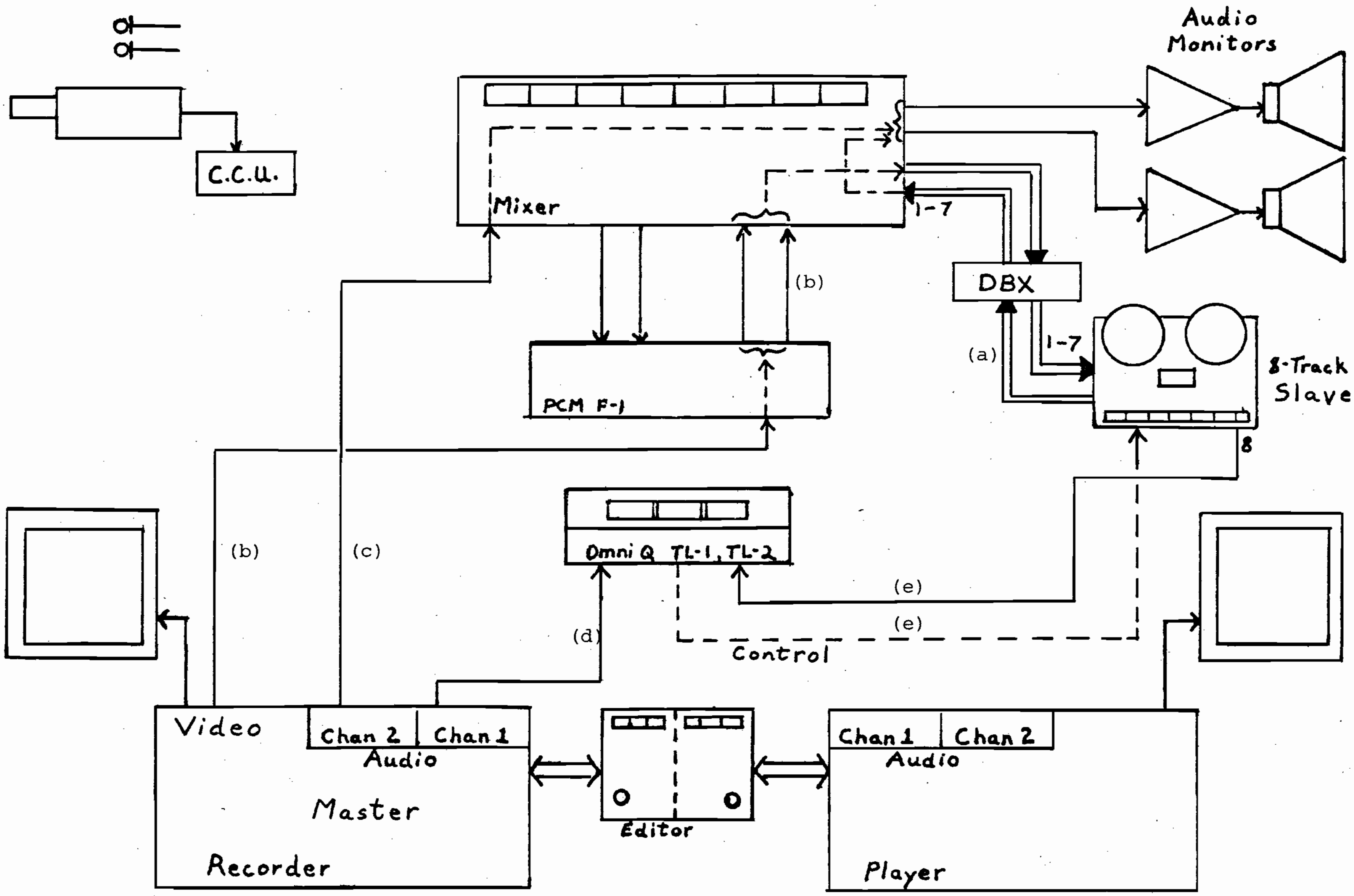


Diagram 3 . Bounce from PCM Audio to Multi-track Analog.

(b) The video output is routed from the video deck with the PCM material to the decode side of the PCM processor. From the PCM processor, the stereo audio is routed to the mixer.

(c) Channel 2 (the analog cue track) is routed to the mixer to be easily monitored.

(d) Channel 1 of the VCR goes to the Master time code input of the synchronizer.

(e) Channel 8 of the multitrack is routed to the Slave input of the synchronizer, and the Control Output of the synchronizer is routed to the control input of the multitrack.

(f) Settings (one VCR only):

Tracking:	As necessary for PCM playback
Drop Out Compensation:	'OFF'
Playback Sync:	'VIDEO'
8-track:	Should be on 'SYNC' to ensure proper time alignment of new material to time code being read from the 8-track. If the time code is taken from tape on the repro head and the new audio is, of course, on the record head, there will be an inevitable offset.

Diagram 4. Mixdown of Multitrack to PCM Submaster

During the procedure, the multitrack is slaved to the video master (on the Player) to ensure proper speed of playback. The time codes of both video master and multitrack should be identical at this stage, so the time code can be transferred directly from the video master to the new PCM submaster.

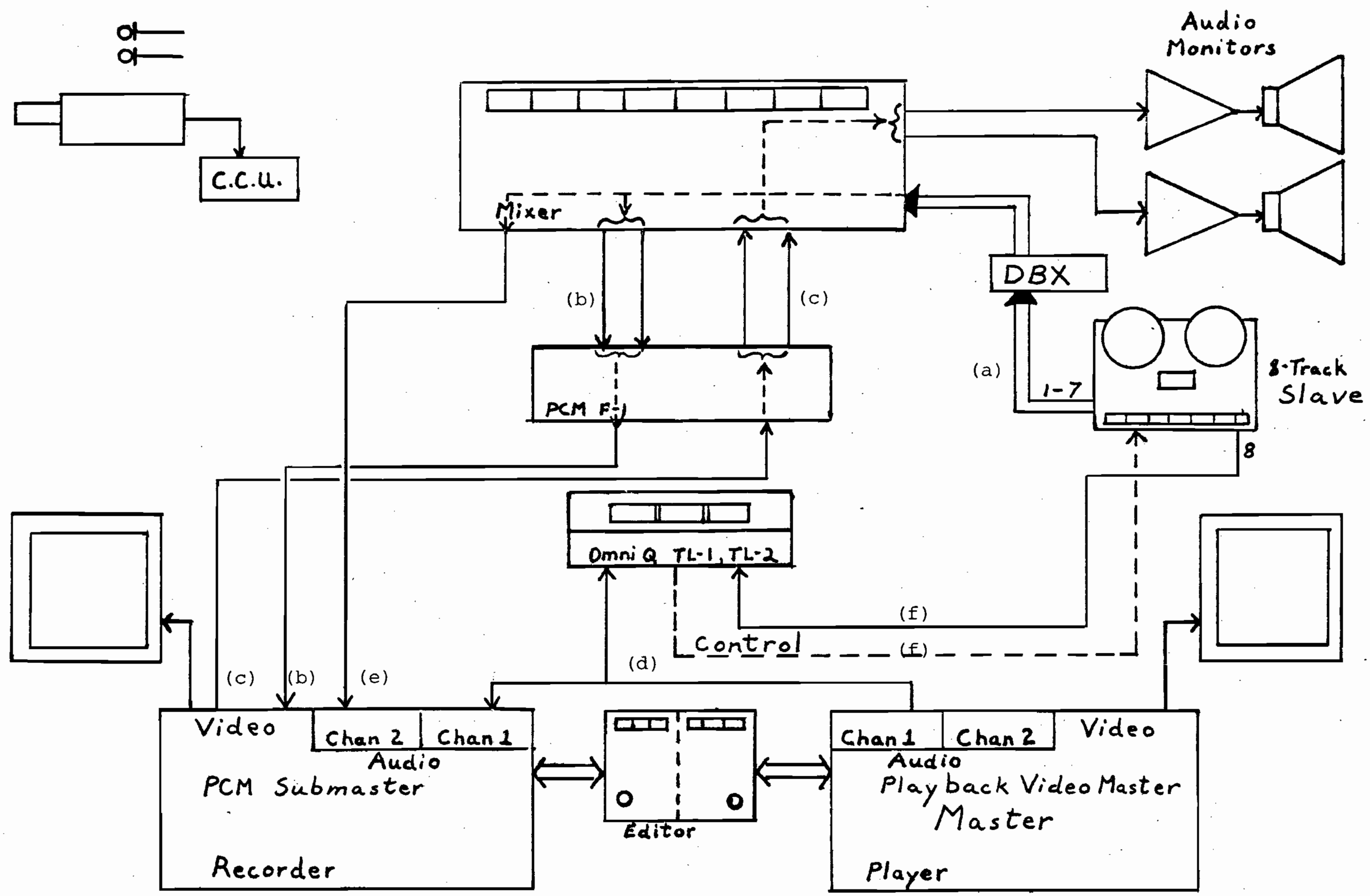


Diagram 4. Mixdown of Multitrack Tape to PCM Submaster under Synchronizer Control

(a) Outputs 1 to 7 of the 8-track are sent to the mixer through dbx decoding.

(b) The mixed two-channel signal is sent to the encode side of the PCM processor, and the video format digital audio signal then goes to the Video Line Input of the Recorder.

(c) Video Output of the Recorder is routed through the decode side of the PCM processor to the mixer for monitoring.

(d) The time code track of the Player (which has the video master on it) goes to the Master Time Code Input of the synchronizer and the channel 1 Input of the Recorder.

(e) A mono mix of the multitrack output is sent to channel 2 Input (analog cue track) of the Recorder.

(f) Channel 8 of the multitrack (that has the pre-striped time code) is sent to the Slave Time Code Input of the synchronizer and the Control Output Sync returned to the 8-track Control Input. Thus, the 8-track is slaved to the video master during the mix.

(g) Settings:

Drop Out Compensation:	'ON' on Player for video master playback 'OFF' on Recorder for recording of PCM submaster
Playback Sync:	'VIDEO' on Player and Recorder
Tracking:	'reset' on both VCRs
8-track:	Reproduction should be from whichever head has the highest audio quality. All audio tracks to be mixed and time code playback must be from the same tape head.

Diagram 5. Reprocessing PCM Audio During a Bounce

(a) The time code of the Player is sent to the synchronizer for display and routed to Channel 1 Input of the Recorder.

(b) The Video Output of the Player (containing the original PCM recording) is sent to the decode side of the PCM processor, and the output stereo analog signal is routed to the mixer.

(c) In the mixer, the PCM output is modified in gain or timbre while being monitored. Note: it is also possible to add other signals while the original signal is in analog form in the mixer.

(d) The reprocessed signal is routed to the encode side of the PCM processor, and the digital audio signal goes to the Video Line Input of the Recorder.

(e) The analog cue track (Channel 2) of the Player is sent to the mixer for monitoring if necessary.

(f) The analog cue track of the Recorder is shown coming directly from the Player, but may also be a mono mix from the mixer.

(g) Settings:

Drop Out Compensation:	'OFF' on both VCRs
Tracking:	As necessary for PCM playback on Player
	'reset' on Recorder during recording
Playback Sync:	'VIDEO' on both decks

Once the bounce is accomplished, it is easiest to transfer the new submaster to the Player for checking. Otherwise, the video input to the PCM decoder must be transferred from the Player to the Recorder VCR.

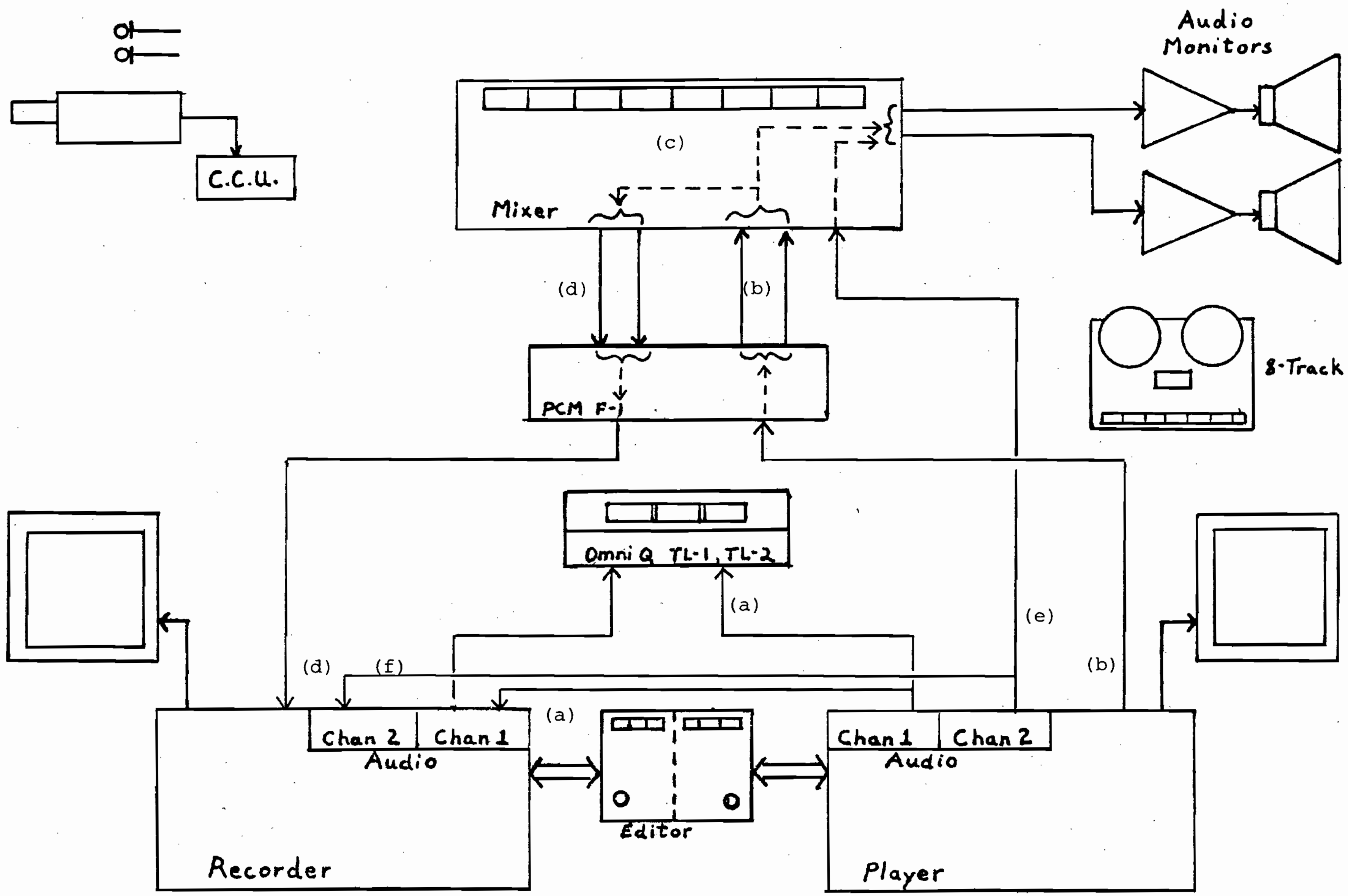


Diagram 5. Reprocessing PCM Audio During a Bounce.

APPENDIX II

SYNCHRONIZERS

This appendix will cover details of the Omni Q TL1 synchronizer and TL2 display module and their use in this application. General use of SMPTE standard synchronization systems will also be discussed, since these are more commonly found in use. Differences between the Omni Q and SMPTE standards will be mentioned. Included and referred to in this appendix are excerpts from the operations manual for the Omni Q TL1 and TL2 modules.

A. <u>Description of the Omni Q Synchronization System</u>

The Omni Q TL1 module contains, first of all, a time code generator which produces a digital code indicating minutes (0 to 59), seconds, and frames. These data are output to tape through a frequency shift keying modulator. The frequency shift keying (F.S.K.) technique is the same as that used to output data from a micro-computer to audio cassette tape.

With F.S.K. two output frequencies are used; one which indicates a '0' bit of binary data and another which indicates a '1' bit of binary data. In the Omni Q system there are two frequency ranges available for these data frequencies to be within. The first, intended for audio tape recorder (ATR) applications, uses a range around 20 kHz. This enables one to record limited bandwidth audio (to 15 kHz) such as a bass on the same track as the F.S.K. time code. Circuitry is provided within the TL1 module to separate the time code from any music signal upon playback of the track. This does provide the advantage of not taking up a recorder channel if you have audio signal that can be recorded while you are recording the time code. With the procedure used in this report--that is, pre-striping the multi-track with time code--this feature was not used.

The second frequency range available is intended for use with video recorders, and uses a lower frequency range. With the system we were using this was dropped even lower than is usual, so that the frequency changes occurred between one of 8 kHz and one of 10 kHz. The lower frequency range is used for video applications because of the extreme unlikelihood of encountering a VTR with the frequency response to handle the higher range.

In a SMPTE standard time code generator, the F.S.K. technique is not used. Instead, a system is employed which produces what is called a bi-phase mark. We will not cover this aspect in this appendix, but the important feature is that it produces a signal with an even lower frequency range, in the 1.2 kHz to 2.4 kHz area. Due to the lower frequencies involved in the SMPTE code format and the signal limiting and conditioning circuits in the time code record and playback sections of our VCRs, the Omni Q time code was not compatible with the separate SMPTE time code channel available. The Omni Q time code must therefore be recorded on one of the two audio channels available. This then limits you to one channel of audio on a tape, and produces the crosstalk interference problem mentioned in section V-E.

At the time of writing this appendix, work is underway to develop a bi-phase modulator which will take the Omni Q data (which is also not to the SMPTE standard) and produce an output which will at least be compatible with the time code channel of the

Sony BVU-200B VCRs which we use in our installation.

To achieve a frame synchronization between two recorders, the TL1 module also contains two time code reading circuits--one for the master, and one for the slave. After demodulation of the F.S.K.-encoded data, the data are sent to a comparator section which compares the values and varies the control output signal which in turn controls the slaved machine. The slaved drive is either slowed or sped up until its code matches that of the master, and subsequently fine control is exercised to keep them the same. It is also possible to have the synchronizer achieve and maintain a fixed difference between the two codes of master and slave by use of the Offset, Offset Increase, and Offset Decrease buttons on the face of the TL1 module.

The nature of the control output signal can be varied by changing an internal interface circuit module. Two basic types of control available are:

1) Varying D.C. Voltage

   This is used for the control of the Teac Model 38 8-track recorder.

2) Varying Frequency Pulse

   This is used to control the speed of the Sony BVU VCRs, which have internal circuits to compare an external frequency reference to internally generated frequencies, and can then vary their speed to match that reference. In our application, the control signal from the Omni Q was applied to the External Playback Sync input available on the Sony BVUs. See Appendix III--Excerpts from the Sony BVU-200B Operators Manual.

It should be noted that both of these controls affect only the normal play speed of the slaved machine. There is no auto-locate function involved, and both machines must be manually cued and started to achieve as close to a sync situation as is possible. If this is not done, it can take a fair amount of time for the synchronizer to achieve the frame sync relationship between the two machines. The speed variance possible is approximately 50 percent for the Teac Model 38 and 10 percent for the Sony BVU-200B.

In order to cue the respective drives it is, of course, necessary to know where they are located. This ability is provided by the Omni Q TL2 module, which contains a display that was used extensively to cue the machines initially and to determine and set edit points. The TL2 module also contains some simple device controllers, but these were not used and will not be covered in this appendix. In the event that the Omni Q system was used for this application, it would be quite feasible to build a display-only module and avoid the extra cost of the complete TL2 module.

The display itself presents a digital readout of minutes, seconds, and frames that may be selected to show the master's time code being played back, the slave's time code being played back, the difference between these two values, or the time code value being generated by the clock within the module. In the event that the time code is no longer received (as when a VCR is stopped), the display holds the last value. This feature, in conjunction with the frame counter of the Sony BVE-500 Edit Controller, is used to determine exact edit locations. This is described in section V-C--How to Determine and Set Edit Points.

It should be noted that the frame counter of this editor has no way of knowing a precise, absolute location on a video tape. It can simply count frames forward or back from any arbitrary zero point which you set. Also, this system of counting pulses from the video control track is quite prone to slippage when reversing directions in search mode and when putting the machine into fast wind modes. It is these problems which make a time code based system a necessity.

The problems that were found with the use of this synchronizer came primarily from its inability to read the recorded time code at very slow speeds or when moving in reverse. Since this is always required when setting edit points, it becomes necessary to use the frame counter and time code reader in conjunction. The basic procedure is to:

1) reset the frame counter to its zero value at a known time code location near the edit point;
2) shuttle at low speed to locate the precise edit point (while this is happening, the time code cannot be read);
3) use the frame counter value to determine where, in relation to the known time code value, the edit point actually is.

B. Other Synchronization Techniques

As with various other sections of this project, we are again getting away with a non-ideal situation. The proper way to synchronize various tape drives should involve a separate sync generator which gives a very stable and accurate time base to the entire system. Sync generators are the rule in more extensive video installations to ensure that all cameras, VTRs, effects devices, etc., are all running in exact synchronization with respect to the horizontal and vertical pulses.

In this case, the sync generator should be providing the time reference to the time code generator itself to ensure that it is really accurate and that a second by the time code is indeed a second. Among other things, this is very good for ensuring the interchangeability of tapes between different facilities. When multiple VCRs are synchronized on replay, the synchronizer should ideally simply get the two machines to the desired frame correspondence and then relinquish control to the external sync signal being produced by the sync generator. Both VCRs, if referenced to this same signal, will then run at exactly the same speed and with their servo mechanisms will maintain a very precise frame relationship; accurate enough, in fact, to enable electronic editing. This prevents the problem of any flutter in the master drive being transferred to the slave by the synchronization system. One other aspect of the sync generator is that the frame rate for color video is not precisely 30 Hz, but is 29.97 Hz. . .. With our present situation, the Omni Q clock being self-timed, a second is a second, but a frame by the time code is not precisely a frame by the video signal. The Omni Q does have an external time base input which will reference the generator to a sync signal and would thereby eliminate this discrepancy.

The case of an audio tape recorder synchronized with a VTR should also have a device controller/synchronizer that will reference to an external sync signal. To maintain sync when the

ATR is the master, it is particularly necessary to reference the ATR to the same sync signal as the VTR will be referenced to when the synchronizer relinquishes capstan control once the frame correspondence is achieved. This is frequently referred to as 'resolving' the audio recorder, and is also beneficial simply to ensure that the ATR is always running at the correct speed.

Synchronization systems vary in capability from the Omni Q TL1, which will simply achieve the desired frame relationship and requires manual cueing and operation of the machines, to systems with complete remote control of all functions, auto-locate, etc. The latter enables pre-programming of extensive edit and special effects lists and interface to audio console automation systems. In our application, the ability to pre-program an edit by time code locations would have made matching audio and video edits very easy to accomplish. Also, due to the nature of the SMPTE code output, it may be read in reverse and in many systems a time code reader will be able to use tach and servo pulses to keep track of the frame locations at very low speeds. With that ability, it is no longer necessary to use the frame counter on the editing unit to determine the location of edit points. Basically, the more remote control and automated functions available on a synchronization system, the easier and less time-consuming the process will become. However, all of these features are not necessary to actually produce a high quality digital audio master for video.

C. Technical Data

The following are excerpts from the Operating Manual for Commerical Electronics' Omni Q TL1 Synchronizer and TL2 Display Module. Used by permission of Commercial Electronics, 1335 Burrard Street, Vancouver, B.C.

3.0 The Omni Q Time-Code

The **Omni Q** time-code is a special digital code that has been developed specifically for your synchronizer. **It is a totally unique code, and is not compatible with other time-codes such as SMPTE.**

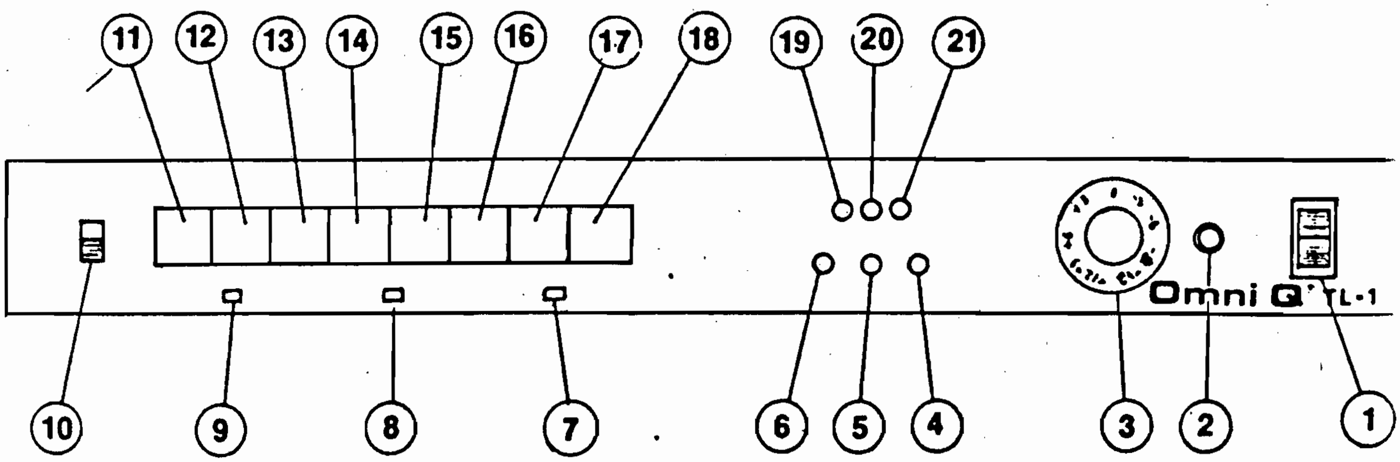
It is recorded continuously and assigns a unique time to any point on the tape in Minutes, Seconds, and Frames. A Frame is normally 1/30 of a second (33.3 ms) but can be changed to 1/25 second (40 ms) for film work or European use.

Unlike SMPTE, and other codes, the Omni Q code incorporates a **check sum parity**. This means that a unique number is generated by the TL 1 based on the time-code data. This number is recorded onto the tape along with the data. When the TL 1 is reading code, it will again calculate a number **based on the actual data read** and compare it to the number stored on the tape. If they are not identical, the TL 1 assumes an error and relinquishes control of the Slave's servo until it again reads valid data. This prevents disasterous results arising from drop-outs, loss of time-code level, or running out of time-code.

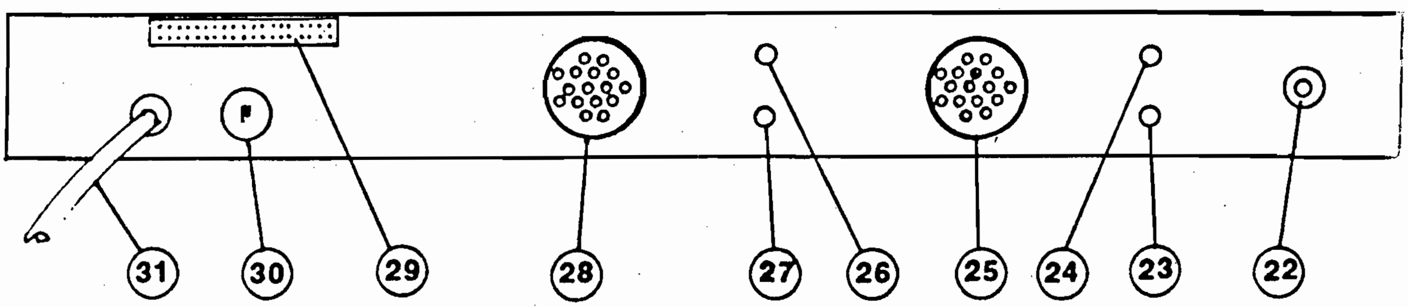
In the Audio Sync Mode the time-code is modulated on a 21 KHz carrier. This enables the recording of other material on the same channel with the following restrictions:

> This material must, obviously, be recorded at the same time as the time-code.
>
> If the material is at an unusually high level, or is highly transient, it may interfere with the time-code. Simple experimentation will determine this.
>
> The time-code channel has a restricted frequency response due to internal filtering. It is shelved 8 dB from 3 KHz to 15 KHz and rolls off from there at 18 dB/octave.

In the Video Sync Mode, the time-code is modulated on a 12 KHz carrier. This is simply because the audio response on a VTR is not sufficient to allow the use of a 21 KHz carrier. When the 12 KHz carrier is used it is not possible to record other information on the same channel.



Front Panel



Rear Panel

4.0 Control Functions

1. **Power Switch**

2. **Calibrate Pot**: Enables you to calibrate the servo output to match your Slave transport's servo circuit.

3. **Phase Control**: Permits a continuously variable offset within one frame (ie: $\pm$33 ms) for phasing and flanging effects.

4. **Slave Status LED**: Indicates that valid time-code is being read from the Slave. Drop-outs on the tape will cause an occasional flicker.

5. **Phase LED**: Indicates that the two devices being controlled are synchronized to within $\pm$50 micro-seconds. Any flickering of this LED is usually an indication of Slave Wow & Flutter.

6. **Master Status LED**: Indicates that valid time-code is being read from the Master. Drop-outs on the tape will cause an occasional flicker.

7. **Slave High-Pass Filter Switch**: Adjusts the Slave carrier filter to match the carrier frequency.

8. **Master High-Pass Filter Switch**: Adjusts the Master carrier filter to match the carrier frequency.

9. **Carrier Frequency Switch**: Selects a carrier frequency of either 21 KHz ("HI") or 12 KHz ("LO").

10. **External Sync Switch**: Selects a high or low frequency input range for an external sync pulse.

11. **External Sync Button**: This latching push-button selects either the internal clock as reference, or an external sync pulse (**IN** position).

12. **Run Button**: This latching push-button starts and stops the time-code generator. The generator will run when the button is **IN**.

13. **Set Button**: This is a momentary push-button. When pushed briefly, this button will jam the last reading taken from the Slave into the generator. When pressed and held for about two seconds, it will set the generator to zero.

4.0 Control Functions **cont'd**

14. <u>**Fast Slew Button**</u>: This latching push-button selects either the Slow Slew Mode or the Fast Slew Mode (button **IN**). The Fast Slew Mode generates a greater control swing over the Slave machine's servo. This is generally used to slew into sync, over long differences, quickly. When running in sync, Slow Slew is used to apply gentle (and, therefore, inaudible) corrections.

15. < <u>**Button**</u>: This momentary push-button allows the Slave to be offset to a position in time that is behind the Master (when in the Offset Mode). A continually increasing offset is created, at a speed determined by the setting of the Fast Slew Button, for as long as the button is depressed. When released, the TL 1 will maintain this offset.

16. <u>**Offset Button**</u>: This latching push-button enables you to enter the Offset Mode (this mode is active when the button is **IN**).

17. > <u>**Button**</u>: This momentary push-button allows the Slave to be offset to a position in time that is ahead of the Master (when in the Offset Mode). A continually increasing offset is created, at a speed determined by the position of the Fast Slew Button, for as long as the button is depressed. When released, the TL 1 will maintain this offset.

18. <u>**Bypass Button**</u>: This latching push-button disconnects the TL 1's servo control from the Slave, allowing it to be free running. This occurs when this button is **IN**. **N.B.: Every time you apply power to the TL 1, the Bypass Button should be depressed and then released!**

19. < <u>**LED**</u>: Indicates that the Slave is lagging behind the Master.

20. = <u>**LED**</u>: Indicates that the two devices are in sync <u>within</u> one frame.

21. > <u>**LED**</u>: Indicates that the Slave is running ahead of the Master.

22. <u>**External Sync Jack**</u>: This input allows an external sync reference to be used. This signal must fall between 20 and 80 Hz.

23. <u>**Master Record Level**</u>: This trim pot permits adjustment of the time-code output level from the TL 1. It is recommended that this be adjusted to yield a reading of -10 VU with the Master's input controls set to where you normally use them.

4.0 Control Functions **cont'd**

24. **Master P/B Level**: This trim pot allows adjustment of the TL 1's input sensitivity. It is adjusted to yield good, steady illumination of the Master Status LED (6).

25. **Master Connector**: Allows connection of the Master device.

26. **Slave P/B Level**: This trim pot allows adjustment of the TL 1's input sensitivity. It is adjusted to yield good, steady illumination of the Slave Status LED (4).

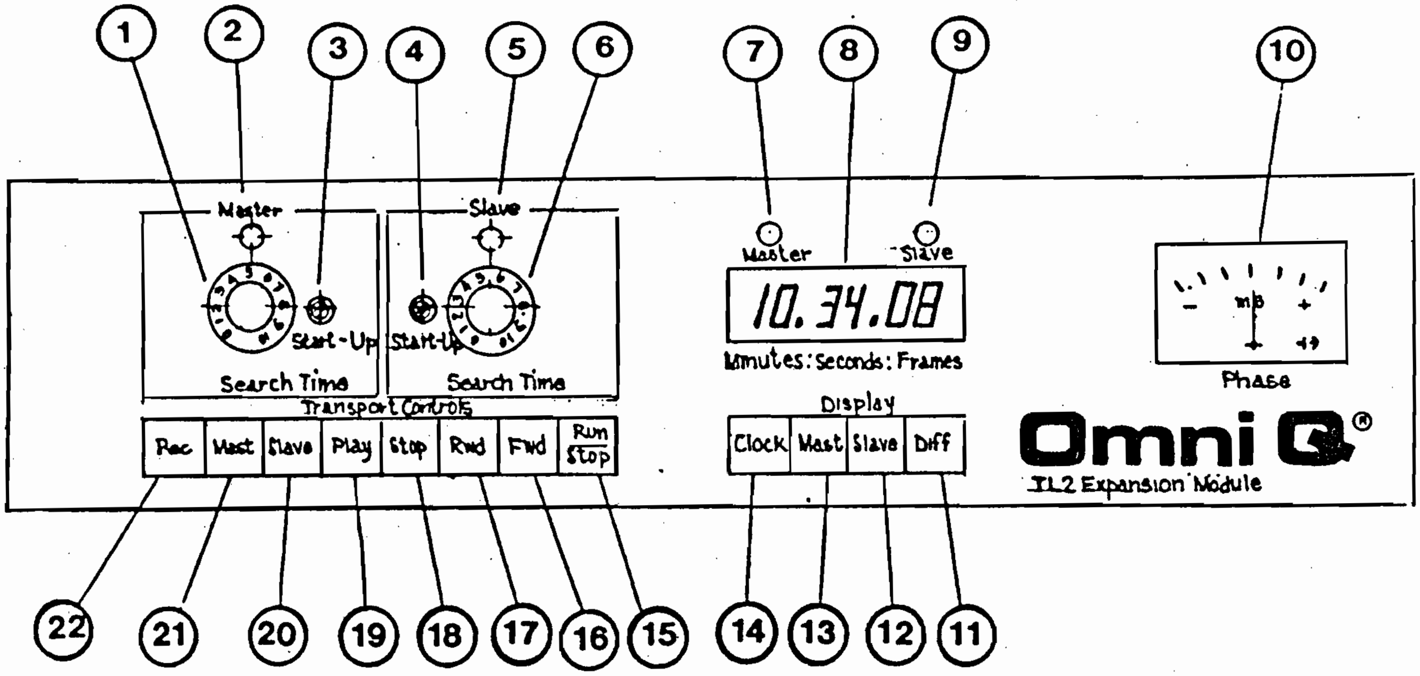
27. **Slave Record Level**: This trim pot allows adjustment of the time-code output level from the TL 1. It is recommended that this be adjusted to yield a reading of -10 VU with the Slave's input controls set to where you normally use them.

28. **Slave Connector**: Allows connection of the Slave machine.

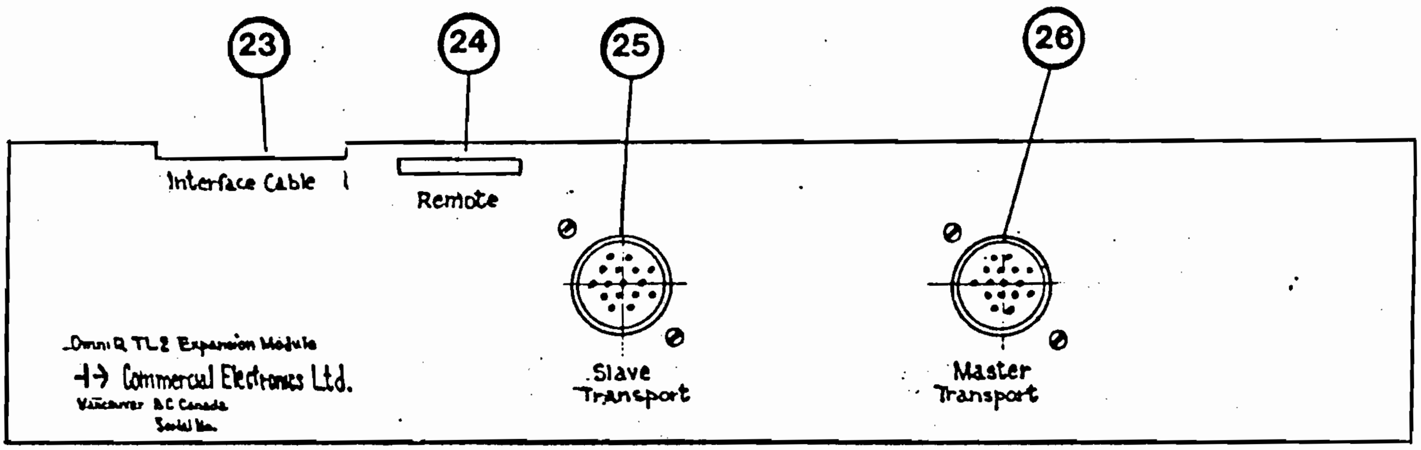
29. **Interface Connector**: Allows connection of the TL 2 Expansion Module.

30. **Power Fuse**: 0.5 A.

31. **Power Cord**: 115 V AC 50/60 Hz.



Front Panel



Rear Panel

Control Functions

1. Master Search Time Adjustment - Enables you to tailor the Locate search time of the TL2 to the characteristics and peculiarities of your master tape recorder.

2. Master Locate LED - Indicates when the master recorder is in the locate mode.

3. Master Start-Up Time Adjustment - Allows you to adjust the delay between the reading of the time-code and the beginning of the search time to compensate for the characteristics of your master tape recorder.

4. Slave Start-UP Time Adjustment - Allows you to adjust the delay between the reading of the time-code and the beginning of the search time to compensate for the characteristics of your slave tape recorder.

5. Slave Locate LED - Indicates when the slave recorder is in the locate mode.

6. Slave Search Time Adjustment - Enables you to tailor the locate search time of the TL2 to the characteristics and peculiarities of your slave tape recorder.

7. Master Display Status LED - Indicates that the display is reading the master providing that the red "Clock" button is not latched in. If both the Master and Slave status LEDs are lit, then the display is reading the difference between the Master and the Slave.

8. Display - Six-digit LCD display indicating Minutes, Seconds, and frames.

9. Slave Display Status LED - Indicates that the Display is reading the Slave. If both the Master and Slave status LEDs are lit, then the Display is reading the difference between the Master and Slave (or the Clock and the Slave).

10. Phase Meter - Gives a visual indication of the TL1's phase control. Each graduation represents 10 mS.

11. Difference Display Button - Enables the display to read the difference between the Master and Slave or the Clock and Slave. Both Master and Slave display status LEDs will light.

12. Slave Display Button - Allows the display to read the slave. The Slave display status LED will light.

13. Master Display Button - Allows the display to read the Master. The Master display status LED will light.

14. Clock Display Button - When latched in, this button enables the Display to read the TL1's clock time in place of the Master.

15. Run/Stop Locate Button - When depressed this button will put any enabled tape machine(s) into the locate mode. If it is latched in, the enabled machine(s) will find the locate time and stop there. If depressed and then released, the enabled machine(s) will find the locate time and BOTH machines will run from that point.

16. Transport Fast Forward Button - Puts any enabled machine(s) into fast forward.

17. Transport Rewind Button - Puts any enabled machine(s) into rewind.

18. Transport Stop Button - Stops any enabled machine(s).

19. Transport Play Button - Puts any enabled machine(s) into play.

20. Slave Enable Button - When latched in, this button enables the slave tape recorder for locate and transport control.

21. Master Enable Button - When latched in, this button enables the Master tape recorder for locate and transport control.

22. Transport Record Button - Puts any enabled machine(s) into record.

23. Interface Cable - Provides the connection between the TL1 and the TL2. N.B.: The TL2 obtains its power from the TL1 and therefore does not have an A.C. cord.

24. Remote Cover - A blank plate to cover a hole provided for a connector for future accessories.

25. Slave Transport Connector - Allows connection to the Slave recorder's transport controls via its remote control socket.

26. Master Transport Connector - Allows connection to the Master recorder's transport controls via its remote control socket.

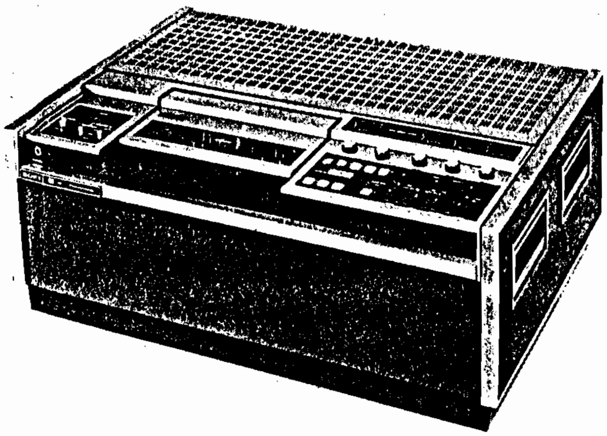
APPENDIX III

Excerpts from the Service Manual for Sony BVU-200B Videocassette Recorder. Used by permission of Sony of Canada.

SONY®

VIDEOCASSETTE RECORDER

BVU-200B



U-matic Professional

OPERATION AND MAINTENANCE MANUAL

SECTION 1
OPERATION

1-1. FEATURES

Easy determination of editing point
When the BVE-500A (optional) automatic editing control unit or the BVR-510A (optional) remote control unit are employed, it is possible to switch the unit's playback pictures from the normal playback mode to reverse playback (or back again) with a one-touch operation. It is also possible to set the tape speed to
x2 double speed (high-speed playback),
x1 (normal playback),
x1/5 (slow-motion playback)
and to monitor the playback pictures for easy determination of the editing point.

Time code recording/playback function
In addition to the conventional video, audio CH-1/CH-2 and control tracks, the tape pattern features a special channel: address track. This allows SMPTE time code to be recorded and played back, without sacrificing an audio channel.

ϕ^2 (Phi Square)-servo loop circuit
This new BVU-200B feature prevents picture disturbances ("flagging", "whipping") at the editing point, since it ensures proper H-phase alignment, as well as frame phase; the H-phase alignment is performed automatically.

High resolution
Comb-filtering is employed for maximum resolution.

Switching during vertical blanking
Video head switchover always occurs during the vertical blanking period.

Capstan servo
The BVU-200B incorporates a capstan servo mechanism which locks onto the external signal (Vertical Drive).

Framing servo
This identifies each odd and even field in a frame, and ensures that edits occur between the exact end of an even field and the exact start of the next odd field, for clean edits.

Insert editing capability
The video, audio CH-1 and audio CH-2 can be edited separately, or in any combination.

Assembly editing capability
Video, audio CH-1 and audio CH-2 can be edited simultaneously.

Time base corrector (TBC) connections
The BVU-200B is provided with an external subcarrier INPUT connector (SC IN) which allows it to be connected to a time base corrector. It is also possible to connect an external dropout compensator (from a TBC, etc.) to the BVU-200B's RF OUTPUT connector.

Automatic/Manual recording systems
This system provides a choice of either AUTO or MANUAL video recording level control. Also, if required, a limiter for audio recording can be selected for virtually distortion-free recording of sudden, excessively strong input signals.

Balanced audio IN/OUT connectors
The audio INPUT and OUTPUT connectors use professional XLR connectors; balanced 600-ohm circuits.

Editing/duplicating connectors
The DUB IN and DUB OUT connectors permit editing or duplicating of the video signals with little degradation for high-quality results, even in multiple generations.

Logic control
The operating mode can be changed without pressing the STOP button between modes, thanks to the use of a logic control system.

Auto stop/auto rewind
This function automatically stops the tape (auto stop) at the end, returns the tape to the start position automatically, and stops it for all modes except rewind (auto rewind).

Servo indicator lamps
These indicator lamps are conveniently located on the front panel, enabling the operator to determine whether the drum, capstan, and framing servos are locked.

Automatic moisture sensor
A condensation sensor inside the head drum assembly activates the AUTO OFF lamp and auto stop function when condensation has formed on the assembly, to prevent possible tape damage.

Power line voltage selector
THE BVU-200B's power line voltage can be set to 100, 120, 220 or 240 V for use anywhere in the world. The main motors feature DC drive; as a result, their operation is unaffected by frequency fluctuations in the power, and the BVU-200B can operate on either 50 Hz or 60 Hz.

1-2. SPECIFICATIONS

MECHANICAL

Weight	46 kg, 102 lb.
Dimensions	645(W) x 277(H) x 463(D) mm 25½(W) x 11(H) x 18¼(D) inch
Tape transport mechanism	U-matic system (3/4-inch cassettes) (accepts small or full-size cassette shells)
Tape speed	9.53 cm/s
Wow/flutter	Less than 0.2% rms
Record/playback time	Maximum of 60 min. with type KCA-60 videocassettes
Fast forward time	Less than 4.5 min. with type KCA-60 videocassettes
Rewind time	Less than 3.5 min. with type KCA-60 videocassettes
Connectors	
AC IN	3-pin AC connector
VIDEO IN x 2	BNC connectors
VIDEO OUT x 2	BNC connectors
AUDIO IN CH-1/CH-2	XLR female connectors
AUDIO OUT CH-1/CH-2	XLR male connectors
TIME CODE IN	RCA phono jack
TIME CODE OUT	RCA phono jack
DUB IN	7-pin male connector
DUB OUT	7-pin female connector
SC IN	BNC connector
SYNC IN	BNC connector
RF OUT (OFF TAPE)	BNC connector
TV	8-pin connector
AUTO EDITOR/ REMOTE	36-pin connector
HEADPHONE OUT	JM-60 headphone binaural jack
Operating temperature	+5°C to +40°C
Storage temperature	−20°C to +60°C

ELECTRICAL

Power requirements	AC 100/120/220/240 V ±10%, 48 to 64 Hz
Power consumption	165 W
VIDEO	
Video recording system	Luminance: FM Chroma: SC low-range conversion
Input	NTSC composite video, sync negative, 1.0 Vp-p$^{+1.0}_{-0.5}$ V, 75 ohms, unbalanced
Output	NTSC composite video, sync negative, 1.0 Vp-p ± 0.2 V, 75 ohms, unbalanced
Dubbing input	Luminance signal: 1.0 Vp-p Chroma signal: 0.9 Vp-p
Dubbing output	Luminance signal: 1.0 Vp-p Chroma signal: 0.9 Vp-p
Horizontal resolution	330 lines (monochrome mode) 260 lines (color mode)
Signal-to-noise ratio	Better than 50 dB (monochrome mode) Better than 46 dB (color mode)
AUDIO	
Input (MIC)	−60 dB, 3 k-ohms, balanced (matches 600-ohm microphones)
(LINE)	+4 dB, 10 k-ohms/600 ohms, balanced
Output (LINE)	+4 dB, low impedance, balanced, (600-ohm load permissible)
(HEADPHONE)	−26 dB, 8 ohms load
Distortion	Less than 2.5% (1 kHz reference level)
Frequency response	50 Hz to 15 kHz
Signal-to-noise ratio	48 dB (at 3% distortion level)
TIME CODE input	0 dB ± 6 dB, 10 k-ohms, unbalanced
TIEM CODE output	0 dB ± 3 dB, low impedance, unbalanced
SC input	2 Vp-p ± 1 V, 75 ohms, unbalanced
SYNC input	0.2 Vp-p ~ 5 Vp-p, negative, 75 ohms, unbalanced (1 Vp-p ± 0.2 V with VIDEO input)
RF output (OFF TAPE)	0.5 Vp-p ± 0.1 V, 75 ohms, unbalanced
Accessories supplied	Dust cover . . . . (1) AC power cord . . . . (1) Dubbing cable VDC-3 . . . . (1)

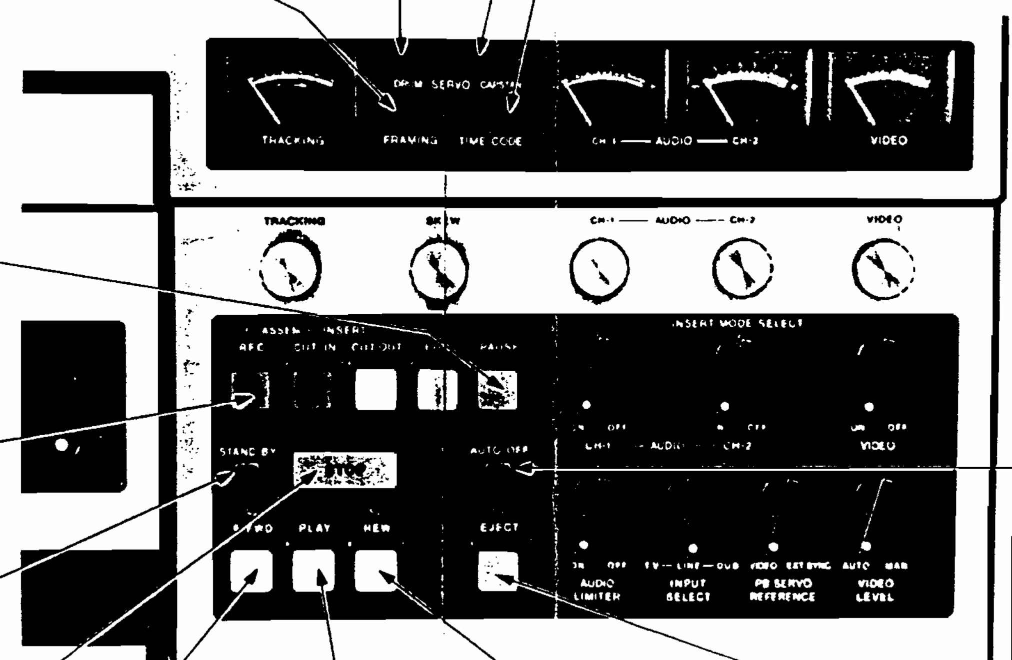
Control Panel (Right)

DRUM SERVO lamp
This lights when the drum servo is locked.

CAPSTAN SERVO lamp
This lights when the capstan servo is locked.

FRAMING SERVO lamp
This lights when the FRAMING SERVO ON/OFF switch is set to ON and the framing servo is activated. (Refer to page 1-10 for details on the framing servo.)

TIME CODE lamp
This lights during recording or playback of the time code signals.



PAUSE button
Depress this button to temporarily stop recording or playback. (The recorder is set to the E-E mode during recording and to the still-picture mode during playback.)
To release, either depress this button a second time or depress the STOP button.

RECORD button
When pressed in conjunction with the PLAY button, the recorder is set to the recording mode after the tape has been loaded (and the STANDBY lamp has lighted).

STANDBY lamp
This lamp lights for about 3 seconds while the tape is being loaded. It also lights when the STOP button is depressed and the tape is ejected.

STOP button
Depressing this button sets the recorder to the stop mode (E-E mode*). E-E mode pictures can also be obtained when the recorder is set to the fast forward or rewind mode.

* E-E (Electronics to Electronics mode)
When the RECORD button is depressed, the video input signal to be recorded is frequency-modulated and then demodulated in the recorder for display on the monitor screen. This condition is called the E-E mode. In this mode, verify that the best possible picture is displayed on the monitor. The audio input signal is simultaneously available at the audio output jack(s). Check the audio signal to be recorded with the machine in this mode.

F FWD button
Depress this button to set the recorder to the fast forward mode.

PLAY button
When this button is depressed, the tape is automatically wrapped around the head drum and the unit is set to the playback mode.

REW button
Depress this button to set the recorder to the rewind mode.

AUTO OFF lamp
This lights when condensation has formed on the head drum assembly or when the auto stop lamp is defective.
Since it takes some time for the moisture sensor to be activated, allow about 10 minutes before switching on the power when the BVU-200B has been brought from cold surroundings into warm surroundings.
After 10 minutes, turn on the power button and check as follows:

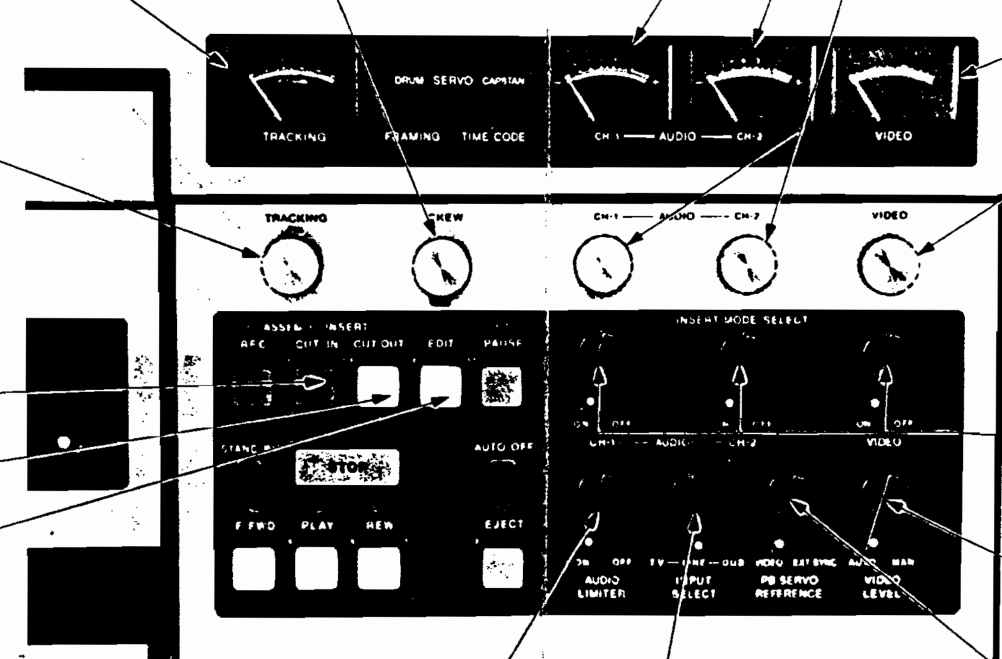
a) If the AUTO OFF lamp does not light, operate the unit normally.
b) If the lamp lights, keep the power supplied to the machine and wait until the moisture evaporates and the lamp goes off. The machine is designed so that it will not operate when this lamp is lighted.

EJECT button
Depressing this button raises the cassette compartment and allows the videocassette to be inserted or removed.

SKEW control
This control mechanically changes the tape back tension. It corrects for diagonal image distortion at the top of the picture caused by individual differences in the tension of various playback tapes. This control automatically returns to its normal position when the unit is set to the recording mode.

TRACKING meter
This indicates the level of the playback FM signals.

TRACKING control
In order for the unit to retain tape compatibility with similar machines, it is possible to adjust the tape tracking during playback. The TRACKING meter indicates the tracking performance during playback. Rotate the TRACKING control to the left or right of center while observing the TRACKING meter. The optimum position is attained when the meter indication is maximum. Always return the control to the center detent position after use.
Note: Servo lock may be lost for a moment if this control is rotated needlessly during recording or editing.



CUT IN button
This button and the EDIT button are depressed simultaneously to start insert editing.

CUT OUT button
Depress this button to stop editing. (The unit returns to the playback mode.)

EDIT (SAFETY) button
Depress the RECORD and EDIT buttons simultaneously at the editing point to start assembly editing.

AUDIO LIMITER selector switch
The limiter control circuit is actuated when this selector switch is turned to ON. The circuit limits sudden surges of input signals to a fixed level during recording. Satisfactory recording characteristics can be obtained with low distortion and a wide dynamic range even when audio levels vary unpredictably.

INPUT SELECT switch
This switch selects the input source which is to be recorded (or edited).
TV: Selects audio and video signals from a connected TV/monitor as the input source.
LINE: Selects audio and video signals connected to the VIDEO IN and AUDIO IN connectors as the input source.
DUB: Set to the DUB position for tape-to-tape editing and duplicating.

AUDIO LEVEL meters (CH-1/CH-2)
These indicate the audio input level when the unit is set to the recording mode. In the playback mode, they indicate the audio output level.

AUDIO LEVEL controls (CH-1/CH-2)
These are used to adjust the audio levels (input/output).
Input: When the unit is set to the stop (E-E) mode, observe the AUDIO LEVEL meters, and rotate the controls so that the meters read zero.
Output: During playback, these controls can be used to adjust the playback level.

VIDEO LEVEL meter
This indicates the video input level.

VIDEO LEVEL control
When the VIDEO LEVEL selector switch described below is set to MAN, this control can be used to adjust the video input level: With the recorder set to the stop (E-E) mode, observe the VIDEO LEVEL meter, rotate the VIDEO LEVEL control until the meter indicator lies within the blue zone (correct level).

INSERT MODE SELECT switches
To perform inset editing, set these switches to the ON position. The BVU-200B will then permit insert editing of the selected signal(s). (They have no effect during playback.) Depending on which signals are enabled for inserting, it is possible to insert just video, just audio, or any combination desired.

VIDEO LEVEL selector switch
AUTO: The sync AGC circuit is actuated and the video signals are automatically recorded at the appropriate level.
MAN: The recording level of the video signals can be adjusted manually.

PB SERVO REFERENCE selector switch
VIDEO: The video input signals are treated as the reference signal and the servo is locked. If all input video signals are disconnected, the unit automatically references to internal sync.
EXT SYNC: The SYNC IN connector input signal is treated as the reference signal and the servo is locked.
This switch works only in the playback mode. During recording, the video input signal becomes the servo reference signal regardless of the position of this switch. (Refer to the Table on page 1-11)

Connector Panel (Rear)

75Ω termination switch
This is the 75-ohm termination switch for the VIDEO IN connectors.
ON: Set here for normal use.
OFF: Set here when one of the video input connectors is to be used as a loop-through output connector.

VIDEO IN connectors (BNC, 1 Vp-p)
These are input connectors for the recording or editing video source. Since these two connectors are wired in parallel, one can be used as a looping output (bridge connection) to other video equipment. When only one connector is used, make sure that the 75 Ω termination switch (see below) is set to ON; when looping through, set OFF.

AUDIO IN LEVEL selector switches (CH-1/CH-2)
These switches select the input level of the AUDIO IN connectors.
HIGH: +4 dB (for line input)
LOW: −60 dB (for mic input)

600 Ω ON/OFF switch (CH-1/CH-2)
The input impedance can be selected for 600 ohms (ON) or 10 k-ohms (OFF) when the AUDIO IN LEVEL switches are set to the HIGH position.

CHROMA LEVEL control
The Y/C mix level of the VIDEO OUT connector playback signal can be varied by turning the control to the left or right of center.

COLOR LOCK control
This recorder contains an automatic phase control (APC) circuit; however, this control can be used to compensate manually for color playback signals which have exceeded the APC range.

TV connector (8 pins)
A color monitor with an 8-pin connector is connected to this connector with the monitor connecting cord.

REMOTE connector (36 pins)
This accepts the Sony BVE-500A editing control unit.

FORCED EXT SYNC switch
OFF: In the playback mode, the servo reference signal is selected in accordance with the setting of the PB SERVO REFERENCE selector switch.
In the record mode, the VIDEO IN connector signal is forcibly selected.
ON: The EXT SYNC IN connector signal is forcibly selected as the reference signal, regardless of the mode of the record and playback, overriding the PB SERVO REFERENCE selector switch. Refer to the table on page 1-11 for further details.

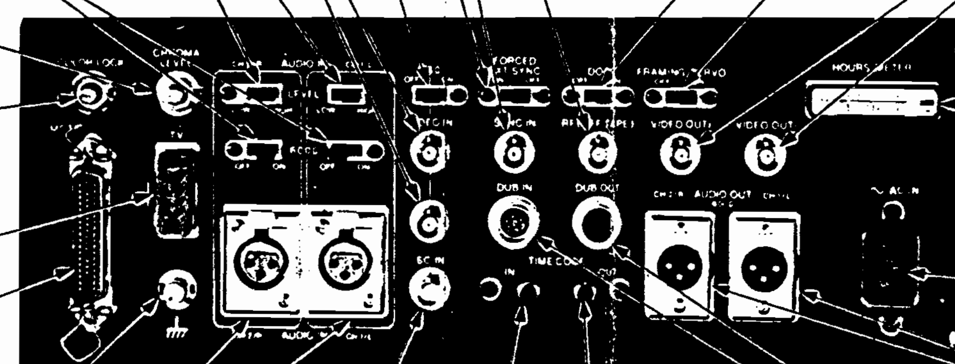
SYNC IN connector (BNC, 4 Vp-p, 75Ω unbalanced)
This is the EXT SYNC signal input connector. It can also be fed a VIDEO signal (1 Vp-p).

RF connector (BNC)
The undemodulated FM signal is fed out here. Connect to an external dropout compensator when unit's built-in DOC is not to be used.

DOC ON/OFF switch
ON: The unit's built-in dropout compensator is activated.
OFF: The unit's built-in dropout compensator is deactivated. Set this switch to OFF when connecting an external dropout compensator or a time base corrector with a built-in dropout compensator.

FRAMING SERVO switch
ON: The framing servo is activated when a VIDEO or EXT SYNC input signal is connected (FRAMING lamp lights). The switch is normally kept at this position.
OFF: When editing a tape recorded with the framing servo deactivated (recorded on a BVU-200A/B with the framing servo off or on an ordinary U-matic recorder), the servo is sometimes disturbed at the editing point. In cases like this, set this switch to OFF to assure smooth video transitions. (FRAMING servo lamp goes off.)

VIDEO OUT connectors (BNC, 1 Vp-p, 75 Ω unbalanced)
These are the unit's video output connectors. Two connectors are provided for simultaneously connecting the video monitor, recorder and time base corrector, etc.



HOURS METER
This meter records the total elapsed time while the unit is in the record, playback or editing modes (max. 1,000 hours).

AC IN connector
The accessory AC power cord is connected here.

Ground terminal

AUDIO IN connectors (CH-1/CH-2) (XLR female)
The audio input signals from microphones or audio equipment are connected to these connectors. Their input level and input impedance are selected by the LEVEL and 600Ω switches above the connectors.

SC IN connector (BNC, 2 Vp-p, 75Ω unbalanced)
This subcarrier input connector is for driving the playback chrominance signal with an external subcarrier (3.58 MHz). A time base corrector is mainly connected to this connector.

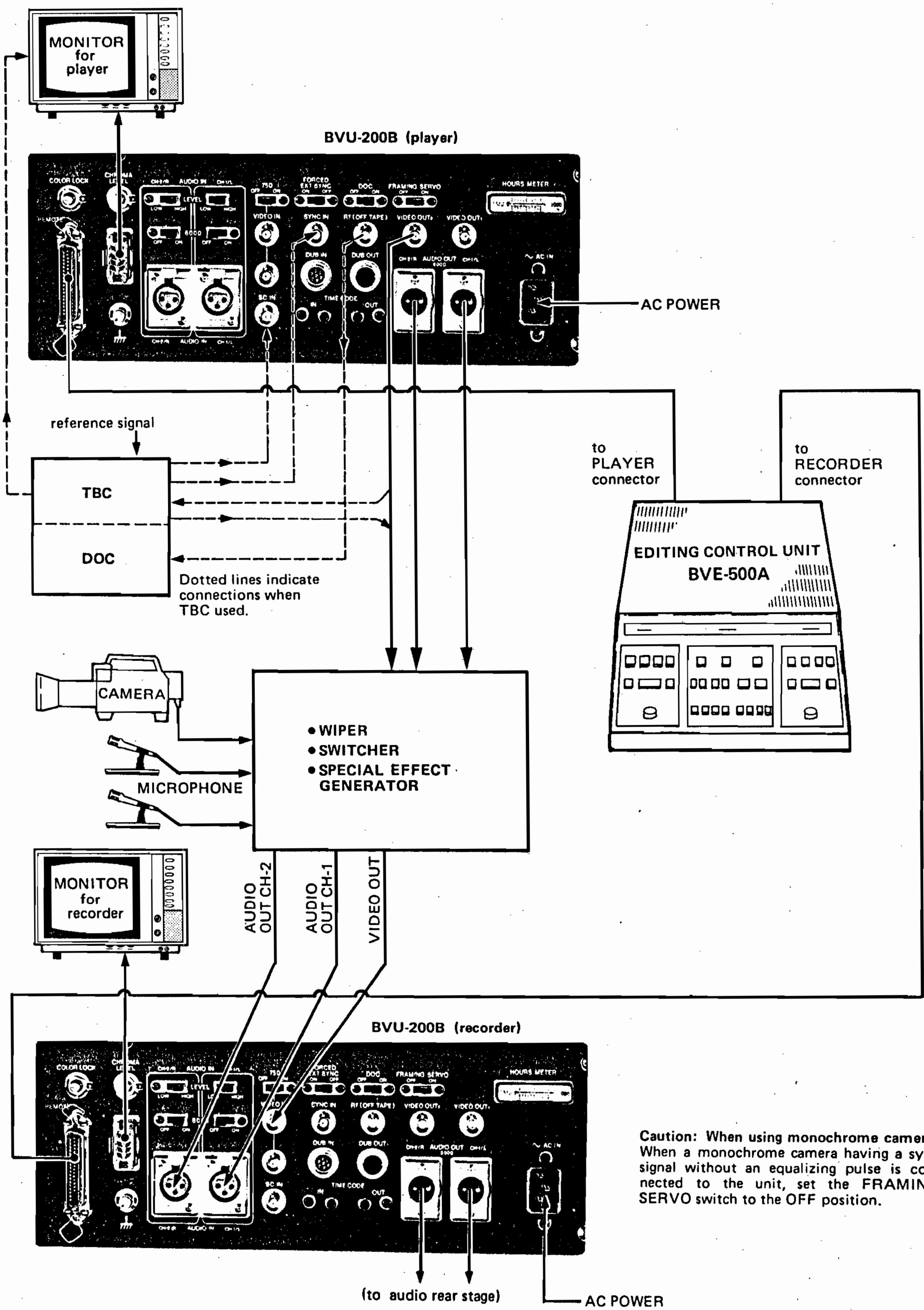
TIME CODE IN jack (RCA phono jack)

TIME CODE OUT jack (RCA phono jack)

AUDIO OUT connectors (CH-1, CH-2) (XLR male, +4 dB)
These are the unit's audio output connectors.
The output signals, whose levels can be adjusted by the AUDIO LEVEL controls, are available at these connectors. The signal at the CH-1 side is selected by the CH-1/L selector switch.

DUB IN/DUB OUT connectors (7-pin connector, IN: male; OUT: female)
By sending video signals from the player to the recorder along the accessory dubbing cable, it is possible to dub or edit video signals, realizing excellent picture quality, better than will be achieved by dubbing signals from the usual video outputs to inputs.

1-4. CONNECTIONS



Caution: When using monochrome cameras When a monochrome camera having a sync signal without an equalizing pulse is connected to the unit, set the FRAMING SERVO switch to the OFF position.

1-7. RECORDING/PLAYBACK

Recording adjustments

Video

① Select the input source with the INPUT SELECT switch.

TV: The video and audio signals from the unit connected to the TV (8-pin) connector can be recorded.
(The audio signals are automatically recorded onto the CH-2 track.)

LINE: The video and audio signals from the unit connected to the VIDEO IN and AUDIO IN connectors (CH-1/CH-2) can be recorded.
When the looping output feature is employed, set the 75 Ω termination switch to the OFF position. Otherwise, set it to the ON position.

DUB: Set the switch to this position for tape-to-tape editing. (Refer to page 1-21 for combination applications of the BVU-200.)

② Select the FRAMING SERVO switch.
Set to ON for recording with the framing servo and to OFF for recording without the framing servo. (It has no effect in playback.)

③ Select the VIDEO LEVEL selector switch to the desired position.

AUTO: The sync AGC* circuit is activated and the recording level is controlled automatically. There is no need to adjust the input level manually.

* Sync AGC: This detects the input signal sync level and automatically controls the level. It is important for the input signal video-to-sync ratio to be at the correct value.

MAN: This allows the recording level to be adjusted manually as follows.

④ With the machine in the stop (E-E) mode, observe the VIDEO LEVEL meter, rotate the VIDEO LEVEL control and set it so that the meter indicator is brought within the blue zone (correct level).

Audio

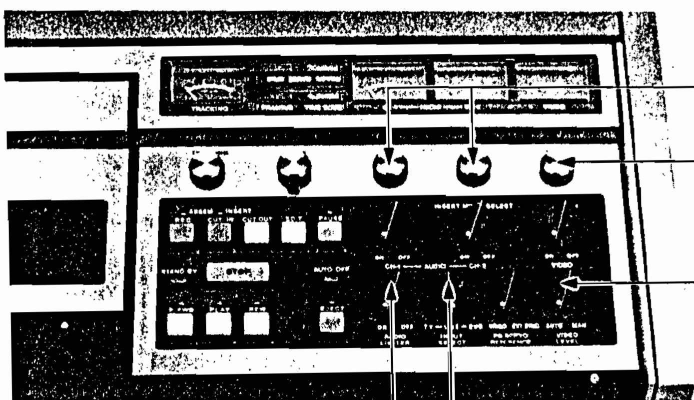
The audio level is adjusted with the unit in the stop (E-E) mode just as with video. There are two audio channels and each must be adjusted individually.

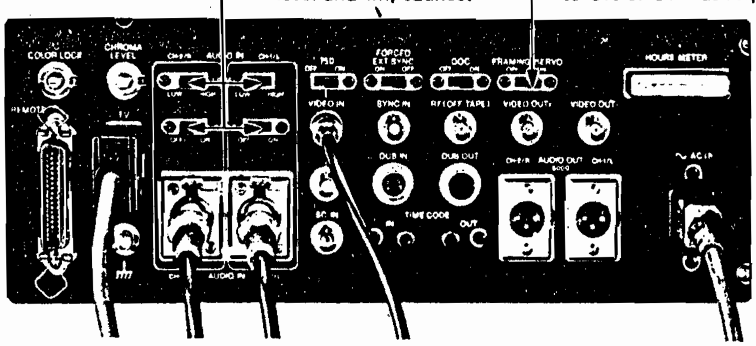
⑤ Set the AUDIO IN LEVEL selector switch (CH-1/CH-2) to the appropriate position.
LOW: −60 dB, 3 k-ohms (with mic connection)
HIGH: +4 dB, 10 k-ohms/600 ohms (with LINE use)

⑥ With the selector switch at the HIGH position, set the 600 Ω ON/OFF switch (CH-1/CH-2), if required, to the appropriate position. When set to ON, the termination impedance is 600 ohms.

⑦ Observe the AUDIO LEVEL meters (CH-1/CH-2), rotate the AUDIO LEVEL controls (CH-1/CH-2), and set for an indication of approximately zero for each meter.

⑧ To record with the limiter, set the AUDIO LIMITER selector switch to ON.

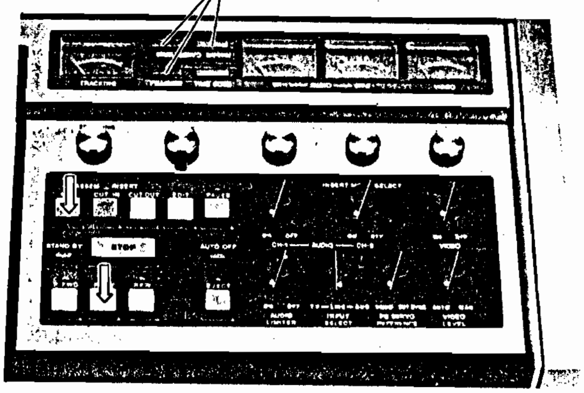




Recording

To simultaneously record the time code, connect the SMPTE time code generator to the TIME CODE IN connectors. Time code recordings automatically involve a limiter and this obviates the need for adjustments.

Press the RECORD and PLAY buttons together.
After the tape has been threaded around the tape drum, the machine is set to the recording mode (the STANDBY lamp lights for about 3 seconds).
This unit comes with indicators which permit the operating state of the servos to be checked. Check that each of the lamps has lighted for servo lock (the machine is operating in a stable state).



RECORD and PLAY buttons depressed together.
(RECORD and PLAY lamps both light.)

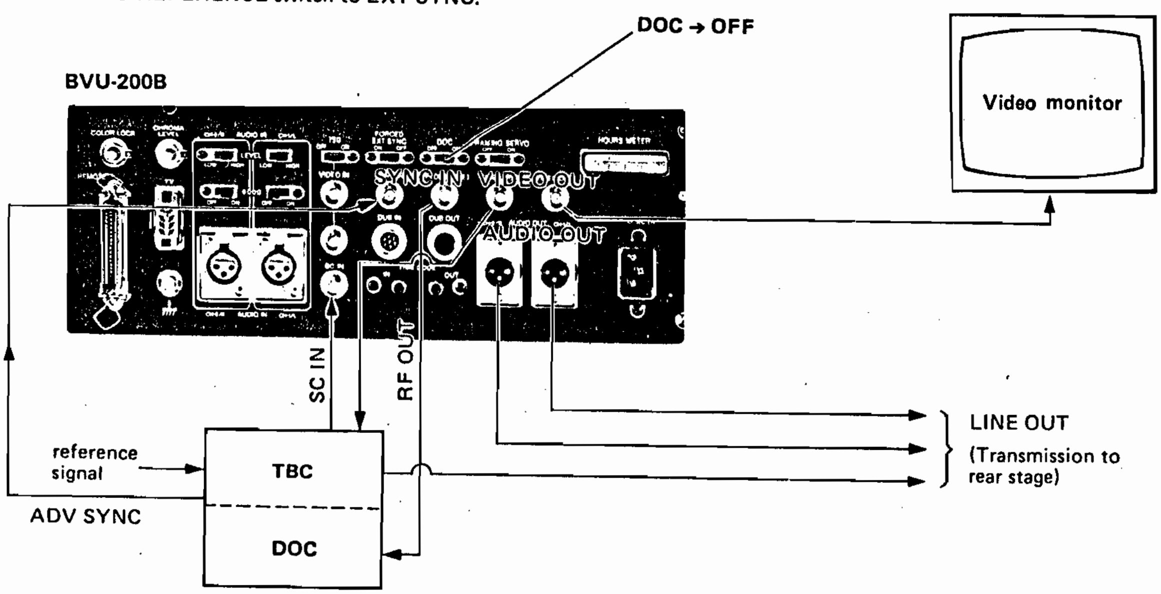
Playback

The machine is set to the playback mode as soon as the **PLAY button is depressed..**
The playback audio level is adjusted by the AUDIO LEVEL controls (CH-1/CH-2) only ("0" indication is the reference). Refer to the figure below when using a time base corrector or dropout compensator.

Precautions during playback
Since this unit is a capstan servo type of recorder (refer to page 1-18), during playback it is necessary to connect a good-quality video signal to the VIDEO IN, DUB IN or TV connector (8-pin), or connect no signal at all. This is because any connected input signal is used as the servo reference.
If a signal source is connected to the VIDEO IN connector during playback, this signal must be regular (i.e., uninterrupted). If noise is included in the signal, the sync will be disturbed and a normal playback picture will not be obtained. (During playback, the input connectors are normally not connected, and in such cases the BVU-200B's internal sync signal generator provides the sync signal.)
If the playback pictures are disturbed, set the INPUT SELECT switch to TV, LINE or DUB so that a position corresponding (or not corresponding) to the connected input signal is selected. If a poor input signal is the cause of the disturbance, the pictures will return to normal in several seconds. As stated, if the quality of a signal connected to a BVU-200B input is in doubt, it is better to disconnect it during playback operations.

Playback using TBC/DOC

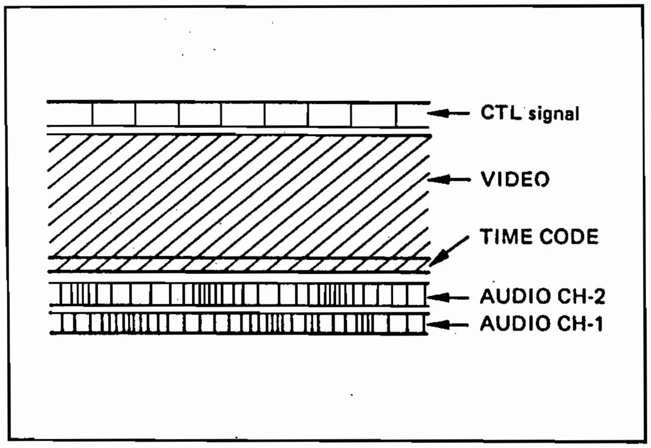
Set the PB SERVO REFERENCE switch to EXT SYNC.



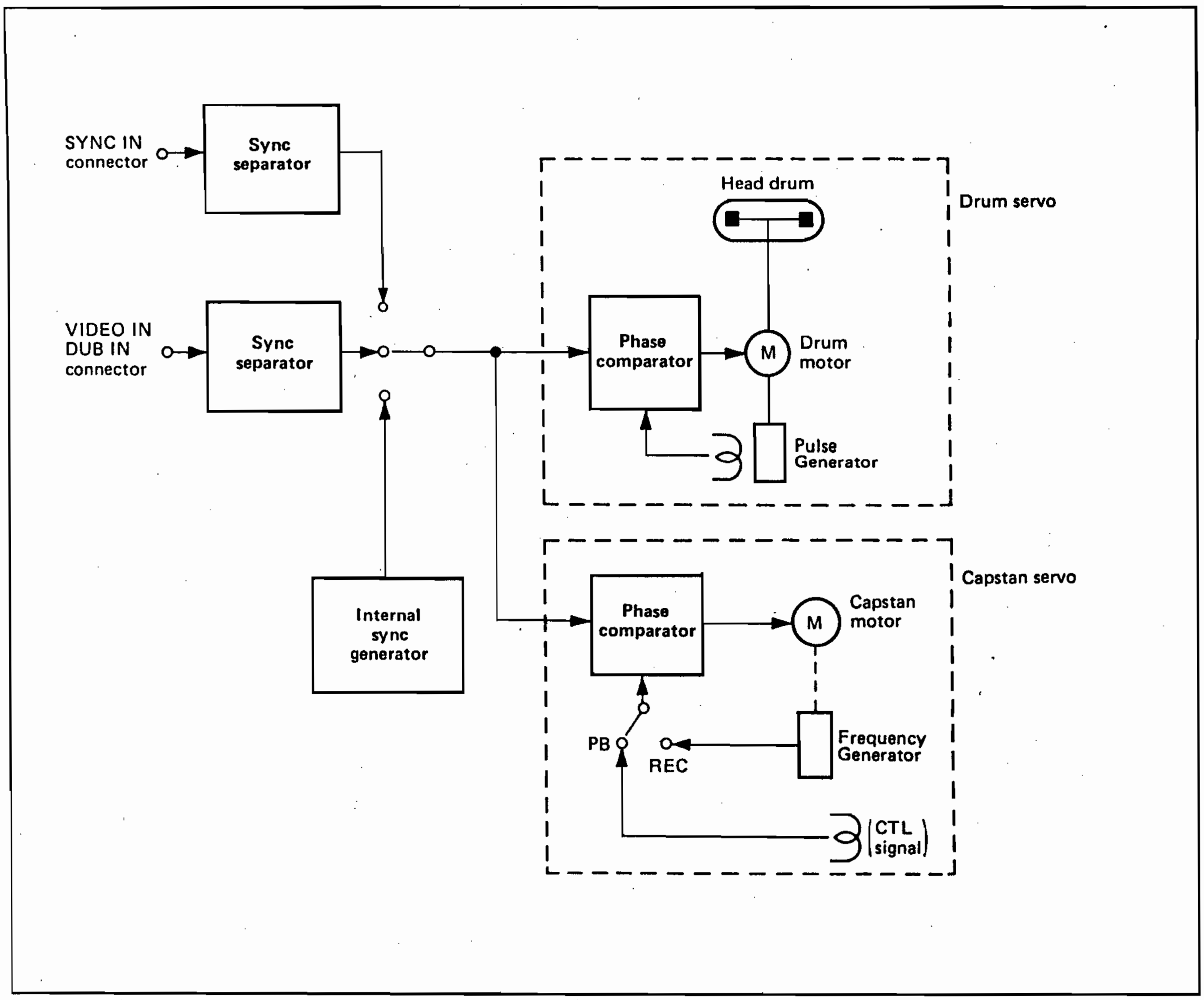
1-8. EDITING

Electronic editing

The figure on the right shows the track pattern of a tape recorded on the BVU-200B. The vertical sync signals in the video signal are recorded at the start (inside the address track) of each video track. This is because during recording the relationship between the video head rotational phase and the vertical sync signal is kept constant. When the head drum goes through a single rotation, two video tracks are recorded by the two video heads mounted on the head drum. During playback it is necessary for the video heads to trace the video track accurately (tracking) and for this purpose the CTL (control) signal is employed. The CTL signal is recorded as a reference signal on the upper edge of the video track, and, during playback, the operating conditions can be made the same as with recording.
It is imperative these conditions be maintained when editing. For this reason, the BVU-200B is equipped with both drum and capstan servos.



Block diagram of servo system



Drum servo

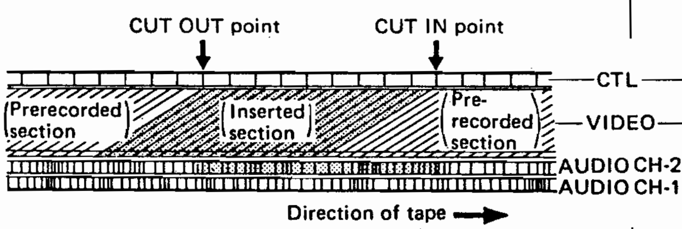
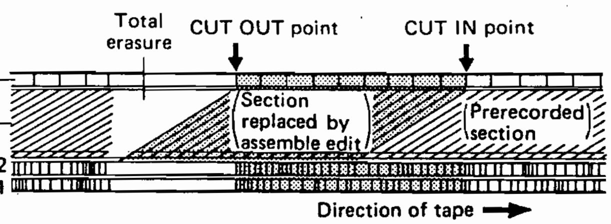
As shown in the figure above, the pulse generator is connected directly to the video head drum shaft.
The phase of the pulse generator output and the phase of the external input signal are compared, the output signal of the phase comparator becomes the motor control signal and this is used to control the rotation of the head drum.
(When there is no external signal, the phase of the internal sync signal generator output and the phase of the pulse generator output are compared and the head drum rotation is controlled.

Capstan servo

The capstan servo controls the rotation of the capstan motor. In the recording mode, the rotation of the capstan motor is controlled by comparing and synchronizing the frequency generator (FG) output signal with the external signal from the VIDEO IN (DUB IN) connector or SYNC IN connector signal (or with the internal sync signal when no external signal is applied).
In the playback mode, for correct tracking the phase of the CTL signal is compared with the phase of the external input signal (or internal sync signal).
When the capstan servo is employed, it is possible to synchronize the external input signal and the playback signal and so there is no disturbance in the pictures at the editing point during the editing operation.
As outlined above, the drum servo causes the head drum to rotate with the video input signal, for editing to be in phase; moreover, the playback tape speed is accurately controlled by the rotation of the capstan, using the capstan servo. By this method, it is possible to switch scenes at the editing point smoothly.

Insert and Assemble Editing

Two forms of editing, insert and assemble, can be performed with this machine.

Insert	Assemble
Insert editing refers to the insertion of new signals, and only those which are required, onto a prerecorded tape.	Assemble editing refers to connecting the signals in sequence one at a time on a single tape.
	
• A prerecorded tape can be played back, signals inserted (VIDEO and AUDIO CH-2 in the above figure) and the previous signals (AUDIO CH-1 in the above figure) can be left intact on the tape. • Even when inserting signals in the middle of a prerecorded tape, the pictures are not disturbed at the cut out point and they are joined smoothly to the previously recorded sections. • In order to synchronize the input signal for insertion with the previously recorded sections (signals which are played back) during insert editing, the tape is made to travel using the recorded CTL signal as the reference signal. Therefore, it is necessary for a continuous CTL signal to have been recorded at the place on the tape where the new signals are to be inserted. The "old" CTL signal is kept intact; it is the one signal that is not replaced in insert editing.	• With assemble editing the CTL, VIDEO and AUDIO CH-1 and CH-2 signals are all replaced by fresh signals as in the above figure. • There is no need to record the CTL signal on the tape beforehand. With assemble editing the vertical sync signal included in the video signal is freshly recorded onto the tape from the start of editing as the CTL signal. • If signals are assembled in the middle of a prerecorded tape, the pictures will be disturbed at the cut out point and there will be no smooth connection with the prerecorded sections (due to total erasure). • For a smooth connection between the pictures, the signals must be assembled successively with the section of tape just before the cut out point serving as the cut in point for the next signal. Always record a little after the intended cut-out point in order to satisfy this condition.

Insert editing

① Select the mode for insertion using the INSERT MODE SELECT switch. This is an ON/OFF switch and the input source corresponding to the ON position can be inserted. The OFF position is the normal position when using an automatic editing controller (BVE-500A).

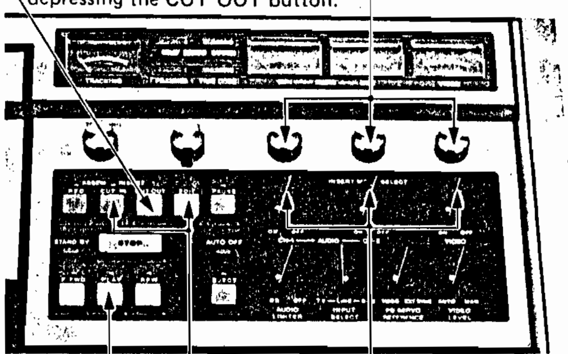
② Adjust the video and audio levels of the material to be inserted. Refer to recording adjustments on page 1-15.

③ Depress the PLAY button to set the machine to the playback mode.
The E-E pictures appear on the monitor for as long as the CUT IN button is depressed when the video INSERT MODE SELECT switch is ON.
However, the E-E pictures will not appear with cassettes whose red cap has been taken out.

④ Once the insert location is reached, depress the CUT IN button and EDIT button together (the start of insert edit).

⑤ When you wish to end the insert edit, press the CUT OUT button and the machine will return to the playback mode.

Note: Time code inserts cannot be made.



Insert editing precautions

Insert editing is a process which uses the CTL signals recorded on the tape, controls the tape speed and edits the video and audio signals in this state. Therefore, it is **necessary to continuously record the CTL signal up to about 5 seconds before and after the insert location as well as from beginning to end.** Remember that the capstan servo will be disturbed and accurate editing will no longer be obtainable if the intervals between the CTL signals are irregular or if noise is included in the video signal (composite video signals including the vertical sync signals) from an external source which becomes the reference signal.

Recording the control signal

Normally, prerecorded tapes contain the CTL signals with which can be video inserts made. When insert editing onto a blank tape, however, it is necessary to record the CTL (control track) signal beforehand onto the tape. This can be performed in any of the following two ways. Be sure to connect the video signal to the VIDEO IN connector.

- Connect a video camera which has its lens capped, and record its sync signal.
- Connect a standard video signal generator which has been set to produce black, and record its output signal continuously.

In all cases, completely record the blank cassette with CTL signals before editing.

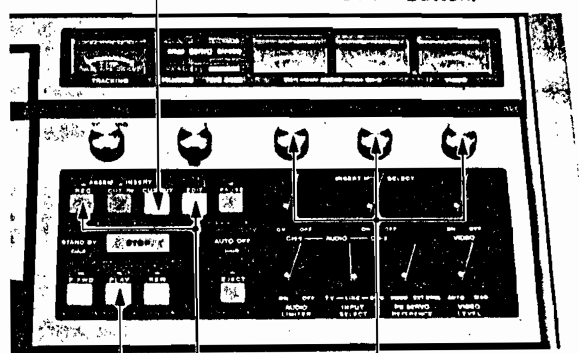
Assemble editing

① Adjust the audio and video levels of the material to be assembled. Refer to recording adjustments on page 1-15.

② Depress the PLAY button and set the machine to the playback mode. E-E pictures will appear on the monitor for as long as the RECORD button is depressed but they will not appear with cassettes whose red cap has been taken out.

③ Once the assembly location has been reached, depress the RECORD and EDIT buttons together (this starts the actual assembly edit).

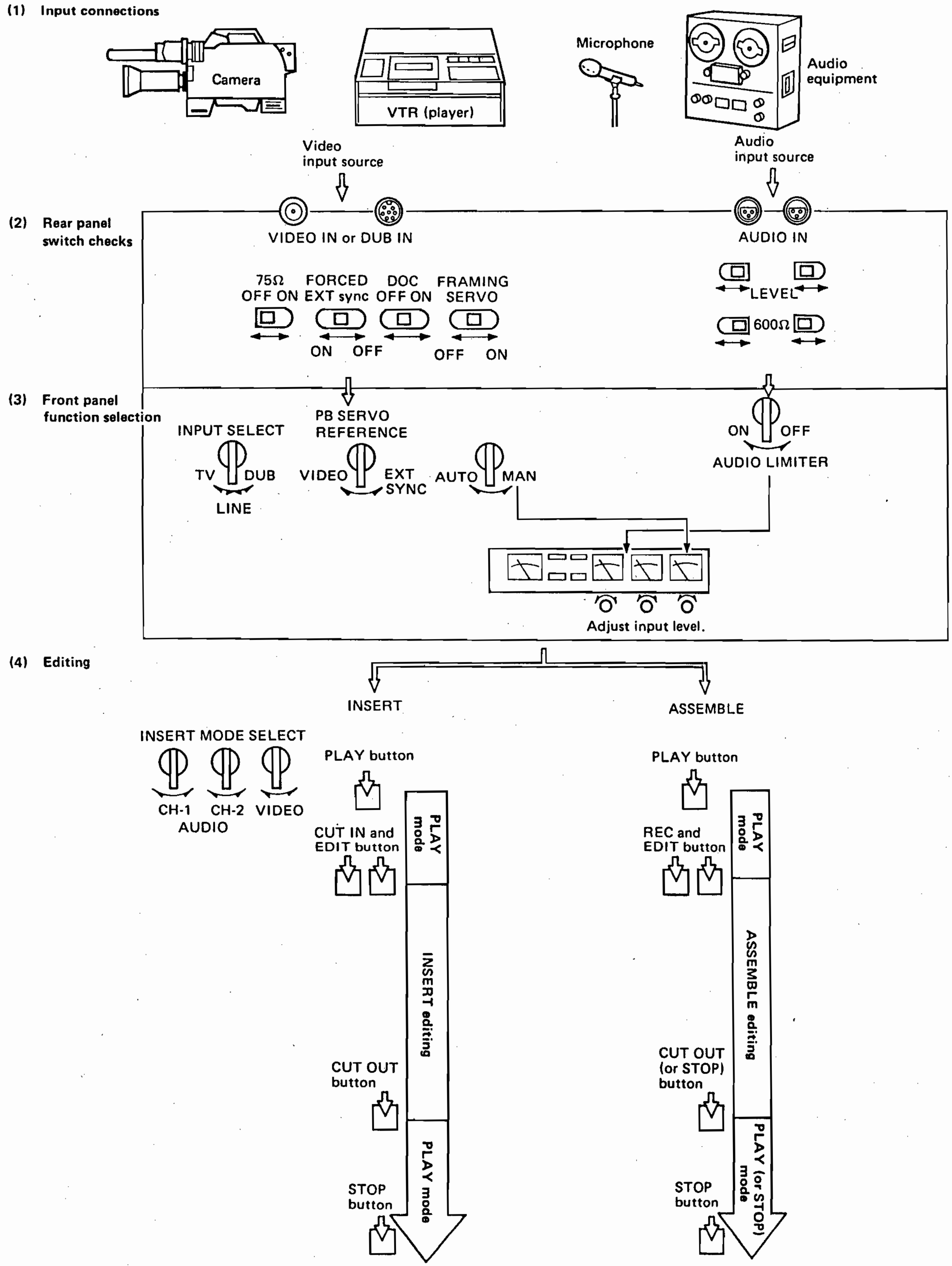
④ When the assembly is completed, depress the CUT OUT button (in which case the machine is returned to the playback mode) or the STOP button (in which case the machine is set to the stop mode).



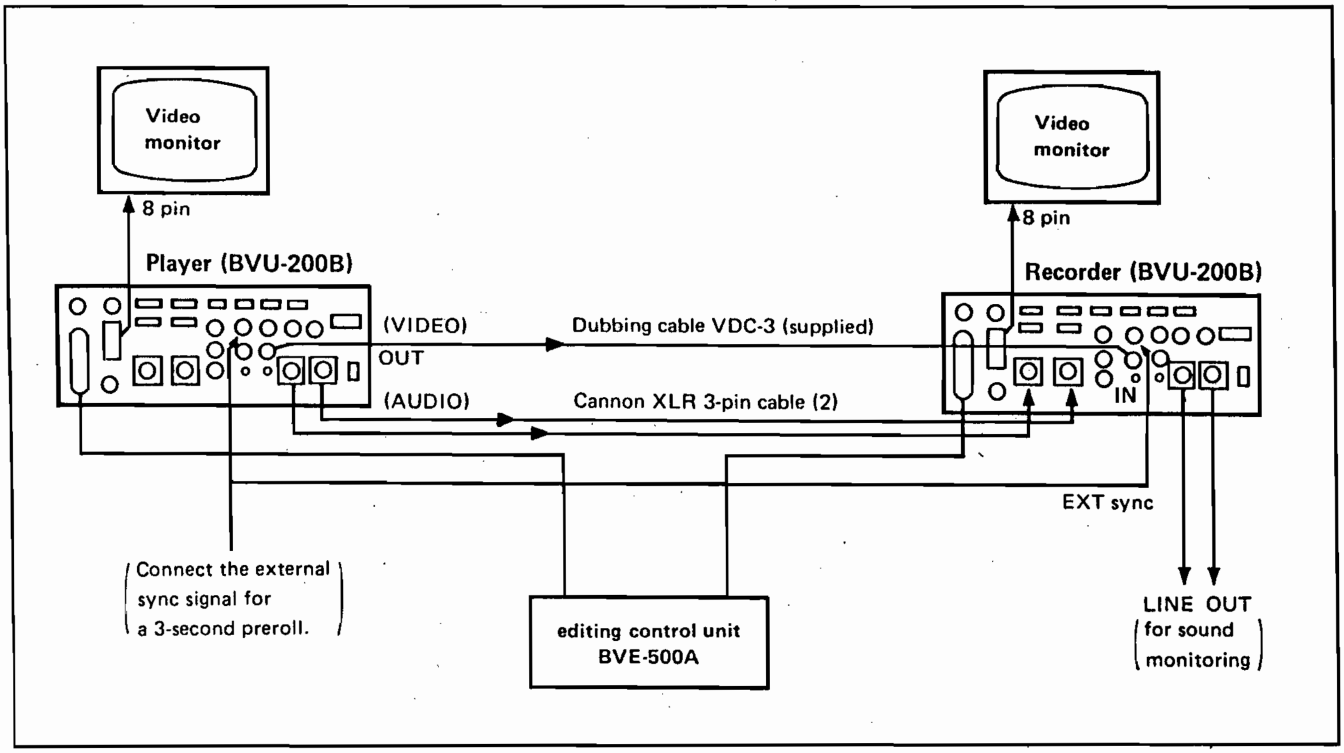
Editing precautions

- Adjust the tracking and skew of the player before editing.
- Under no circumstances should the tracking and skew controls on the player or recorder be moved during editing.
- Proceed with editing after having checked that the playback pictures have stabilized. (Check the servo lamps.) If editing is performed with unstable pictures, the pictures will not be joined cleanly.
- If a tape which has been recorded with the framing servo disengaged (recorded with the BVU-200B's FRAMING switch at OFF or recorded on an ordinary VTR) is edited, the servo may be disturbed at the editing point. In cases like this, set the FRAMING switch to OFF.
- The PAUSE button is designed to temporarily stop the tape during recording and playback. If it is used to start editing suddenly from the pause mode, the pictures will be disturbed at the cut in point. The PAUSE button can not be used to freeze ordinary pictures and record them on another tape.

Editing Procedure

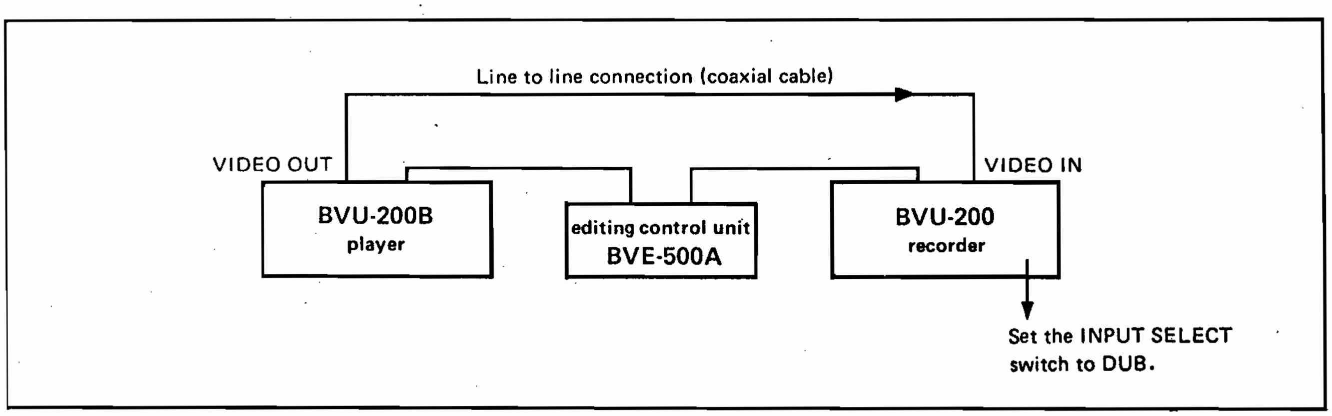


Tape to Tape Editing Precautions



The basic configuration for tape to tape editing using the DUB IN and DUB OUT connectors is shown in the figure above.

- Connect the DUB OUT connector of the player to the DUB IN connector of the recorder. (Avoid using two dubbing cable connections.)
- The BVE-500 must be modified slightly for a BVU-200B/BVE-500/BVU-200B configuration. Contact your nearest Sony service center for details on the modification.
- When editing with a line to line connection using a BVU-200/BVE-500A/BVU-200B configuration, **use the BVU-200B as the recorder**, as in the figure below, and **set the INPUT SELECT switch to DUB.**
  This will allow the benefits of the dubbing function to be realized.
- When editing using a BVU-200A/BVE-500A/BVU-200B configuration, use the BVU-200B as the recorder. The instantaneous transition of pictures at the editing point is improved, due to the action of the BVU-200B's ϕ^2 servo loop circuit.
- If the tape to be edited has been recorded on a VTR without a framing servo, or recorded with the FRAMING switch at OFF, "whip" (a short-term disturbance at the top of the screen) may result at the editing point.
- "Whip" may also result at the editing point when a 3-second preroll is used; if so, use the normal 5-second preroll instead.



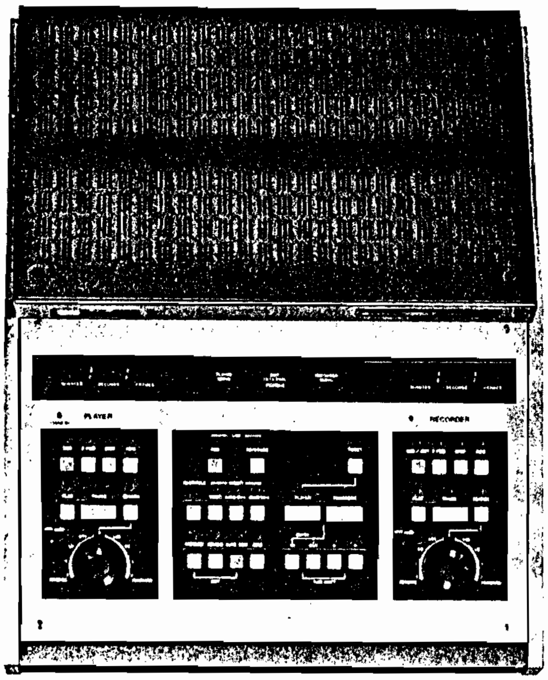
APPENDIX IV

Excerpts from the Service Manual for the Sony BVE-500A Automatic Editing Control Unit. Used by permission of Sony of Canada.

SONY®

AUTOMATIC EDITING CONTROL UNIT

BVE-500A



U-matic Professional

OPERATION AND MAINTENANCE MANUAL

SECTION 1
OPERATION

1-1. FEATURES

Search dial type control

The Search Dial permits the tape to be searched "film style" in both directions. This is a "self-return" dial. The tape shuttle speed can be set by using the Lock Lever to hold the dial in any desired position. To release, just turn the dial.

Dual-stage blinking display

Editing is accomplished simply by following the order of the blinking lights above each button. The functions indicated by the lamps are listed below:

* When the recorder is set in the playback mode, the ASSEMBLE and INSERT (VIDEO, AUDIO CH-1, AUDIO CH-2) lamps blink slowly. When the buttons are depressed and one of the function modes selected, the lamps stop blinking and glow constantly.
* Player and recorder IN and OUT lamps:
  Before any entry is made, the lamps blink slowly.
  After entry and up to the completion of editing, the lamps stop blinking and glow constantly.
  During callout (i.e. when checking the previous entry), the lamps blink quickly.
* The PREVIEW and AUTO EDIT lamps blink slowly.
  When the PREVIEW button is depressed, the PREVIEW lamp blinks quickly during the preroll period. ——► The lamp stays on during the PREVIEW mode when the preview picture appears on the screen.
  When both the PREVIEW and AUTO EDIT buttons are depressed simultaneously, the AUTO EDIT lamp blinks quickly (preroll period). ——► The AUTO EDIT lamp lights (AUTO EDIT mode) and the editing picture appears.
* If the OUT point is not entered yet, the END button lamp blinks slowly. When the END button is pressed, the editing process is ended, and the lamp blinks quickly during roll back. When the END lamp goes out, it indicates the end of the operation.

Automatic entry

When the IN lamp is blinking, just depress the PREVIEW button and the IN point will be entered automatically.
(The automatic entry can not be made when the IN lamp is not blinking.)

Automatic OUT point transfer

When the OUT point is entered on the player, then it is automatically transferred to the recorder during editing.

"Butt" edit

After both assembling and inserting, the recorder automatically returns to the OUT position, which then becomes the next IN position. Thus, the next IN point search is made only for the player.

Manual editing

Manual editing can be achieved on the recorder by using the mode select buttons (ASSEMBLE, INSERT). It is also possible to perform manual assembly with the player.

Automatic pause shut-off mechanism

When the machine has been set to the PAUSE mode for more than about 20 minutes without a break, the PAUSE mode is automatically shut off to protect the tape, and the machine enters the STOP mode.

Full time counter

The counter counts the CTL pulses on the tape, and displays from −79 min. 59 sec. 29 frames up to +79 min. 59 sec. 29 frames.
This system allows editing in both directions from any given point on the tape.

Three-second preroll

When the BVU-200A is operated with external sync, the preroll time can be reduced from 5 to 3 seconds just by changing a switch inside the machine.

External connections

The preroll and cue playback functions can be operated by remote control when other hardware is attached to the EXTERNAL INTERFACE connectors. (TTL level)

Cue control

Cue signals are recorded on the audio track and then erased later. Also, the auto pause function is activated 5 seconds before the on-air point when the CUE PB/ERASE button is depressed.

Quick and accurate review

Playback ranges from 3 seconds before the IN point up to 2 seconds after the OUT point can be reviewed.

Editing camera input signals

When a camera is connected to the player, the player signals are transferred directly to the recorder. When the player is set in the STOP mode, automatic editing can be performed using the recorder only.

Logic control

There is no need to depress the STOP button when selecting a different function button.
For example, you can go straight from search to the fast forward or rewind mode.

Free combination of editing modes

The ASSEMBLE mode is used to make complete edits. However, using the INSERT mode, you can choose any combination of VIDEO, AUDIO CH-1 and AUDIO CH-2 inputs in order to edit only portions of the video program. (In such cases, check that all the BVU-200A INSERT switches are set to OFF.) You can also change the combination of insert modes while in the process of automatic editing.

Edit point memory system

This system is designed to store the player's and recorder's IN points and one of the OUT points in the 3-point memory system. The entered point can be called out on the counter's display board. When an OUT point has been called out, the indicator lamp on the player or recorder blinks quickly.

Time shift : modification of EDIT points

The EDIT points can be easily modified in either direction by a simple operation. The points can be shifted in increments of one frame to facilitate very accurate editing.

Editing rehearsals

Rehearsals can be conducted for all the modes including assemble editing, insert editing, tape-to-tape editing and camera editing by depressing the PREVIEW button.

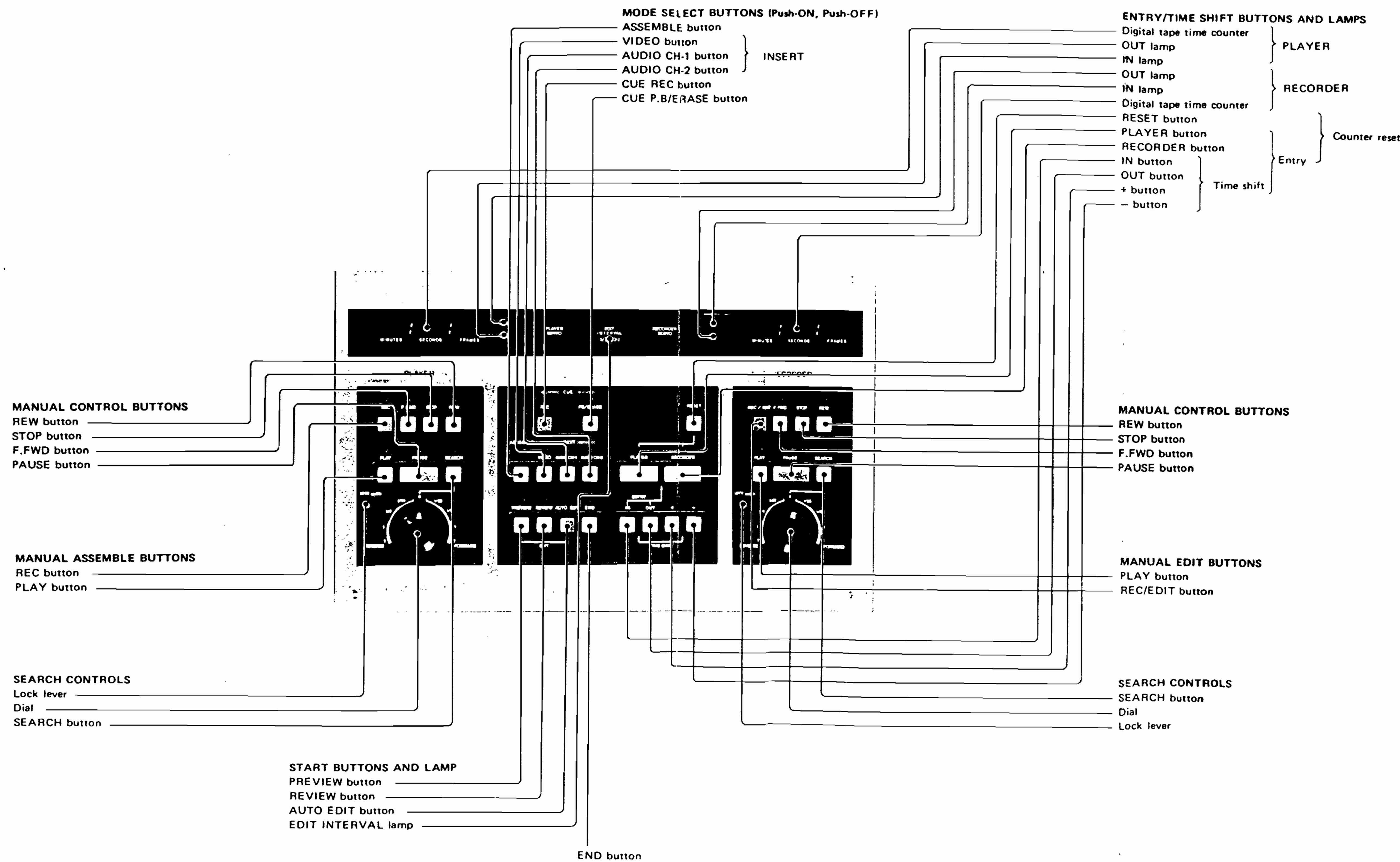
Equipped with adjustable voltage selector

The voltage selector can be switched over to 100 V, 120 V, 220 V or 240 V, whichever meets your local requirements.

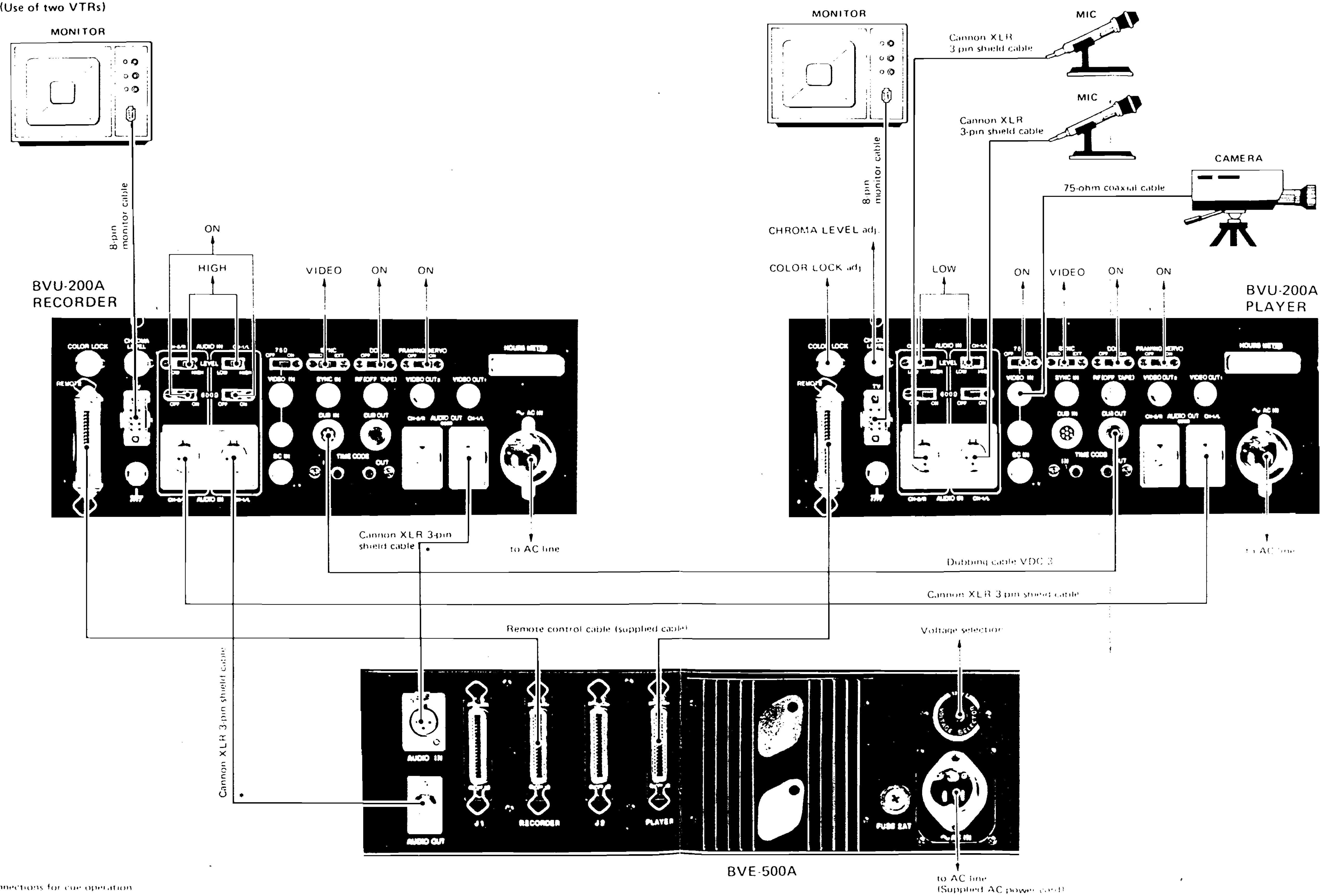
Rack-mounting

The BVE-500A can be mounted in an EIA standard 19-inch rack using the rack-mounting brackets. These are supplied separately.

1-4. LOCATION OF PARTS AND CONTROLS



1-6. CONNECTIONS (1)
(Use of two VTRs)



• Connections for cue operation

APPENDIX V

Excerpts from the Operations Manual for the Sony PCM-F1 Digital Audio Processor. Used by permission of Sony of Canada.

SONY

118.

DIGITAL AUDIO PROCESSOR

PCM-F1

OPERATING INSTRUCTIONS

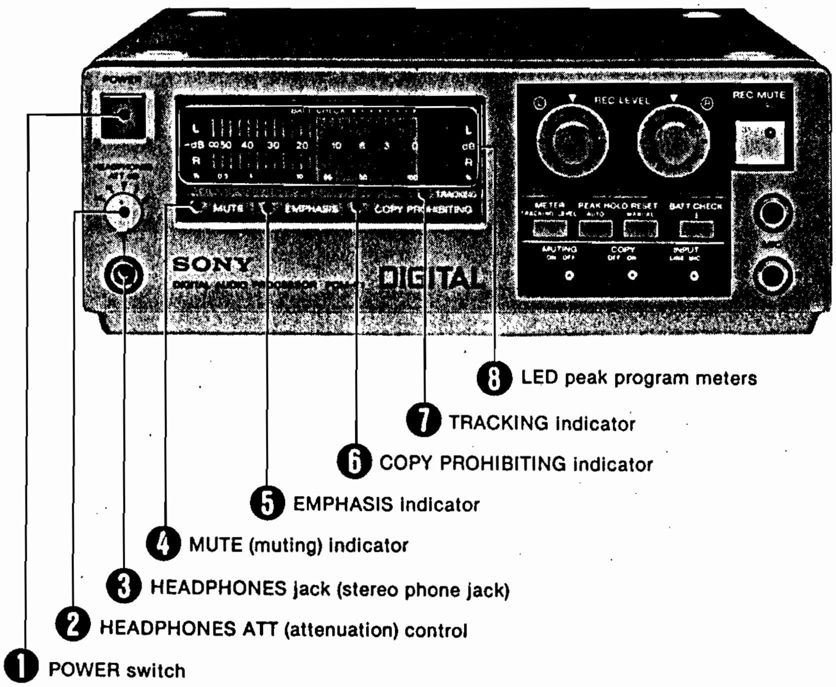
Before operating the unit, please read this manual thoroughly.
This manual should be retained for future reference.

TABLE OF CONTENTS

LOCATION AND FUNCTION OF CONTROLS

Before plugging in or attempting to operate the unit, we suggest that you familiarize yourself with all its switches and controls. Each number in the photo is keyed to the descriptive text.

FRONT PANEL



1 POWER switch
Press to turn on the power. The LED peak program meters will illuminate. To turn the power off, press the switch again.

2 HEADPHONES ATT (attenuation) control
This control adjusts the volume at the headphones. At the "0" position, the rated output is obtained. When this control is set to the "6" position, the level is reduced by 6 dB, and by setting it to "12", "18" or "24", the level is reduced by that amount of decibels from the rated output obtained at the "0" position.

3 HEADPHONES jack (stereo phone jack)
Headphones may be inserted either to monitor the input signals to be recorded or to listen to a recording in the playback mode.

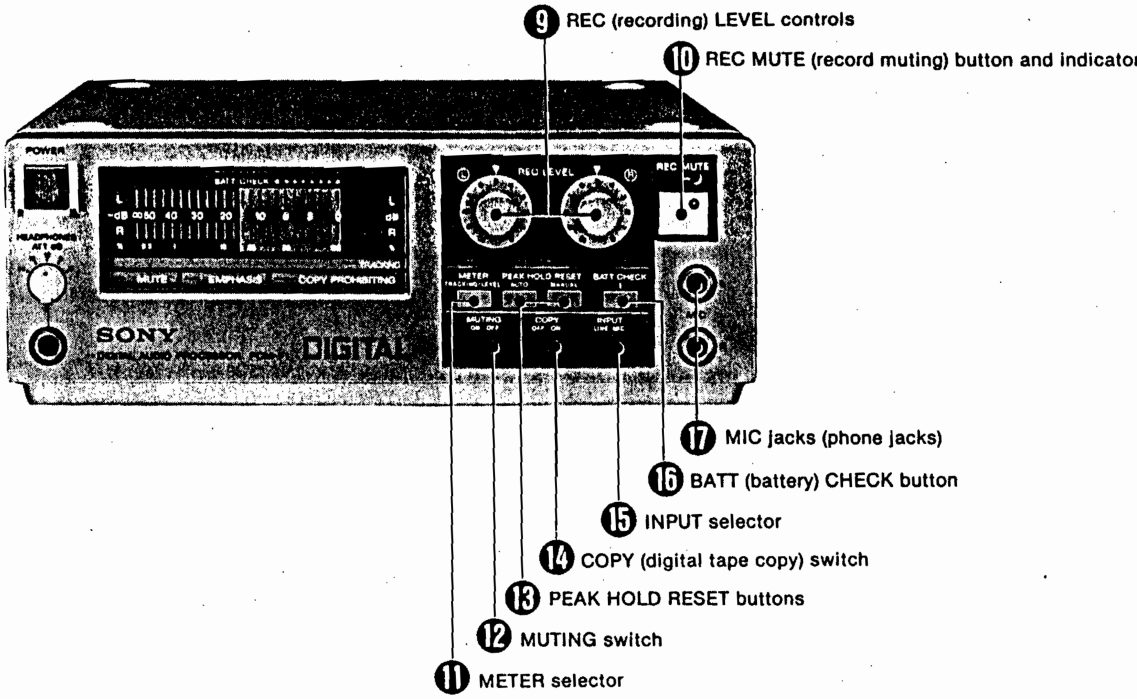
4 MUTE (muting) indicator
If the video cassette recorder is not transporting tape at the proper playback speed (for example, when the tape first begins to move), or if many dropouts occur, this indicator will light up. When the indicator lights up with the MUTING switch set to ON, the muting circuit will activate.

5 EMPHASIS indicator
When recording and playback are made with this unit, the emphasis circuit incorporated in the unit activates during recording (pre-emphasis) and playback (de-emphasis) and the EMPHASIS indicator illuminates.
When a tape recorded without pre-emphasis with a PCM digital audio processor other than this unit is played back with this unit, the EMPHASIS indicator will not illuminate.

6 COPY PROHIBITING indicator
When a tape with a tape copy prohibiting code is played back, this indicator will light up to show that a digital tape copy cannot be made.

7 TRACKING indicator
When you press the METER selector, the lower LED peak program meter will be changed to a tracking meter, and the TRACKING indicator will light up.

8 LED peak program meters
These meters show the peak input level of each channel during recording, and the recorded level during playback. They follow the transient peaks of high-level inputs that are too brief to be followed by conventional VU meters so that the optimum recording level can be accurately set. For easy reading, the meters hold the highest peak while indicating the varying levels lower than the peak.
While the BATT CHECK button is kept depressed, the upper meter for the left (L) channel shows the battery pack condition.
When the METER selector is pressed, the lower meter for the right (R) channel shows the tracking condition of the video cassette recorder.



⑨ REC (recording) LEVEL controls
These controls adjust the recording level. The left knob is for the left channel and the right knob for the right channel.

⑩ REC MUTE (record muting) button and indicator
Keep this button depressed to eliminate unwanted material and to insert a blank space during recording. While the button is kept depressed, the REC MUTE indicator will illuminate.
See "Record muting" on page 13.

⑪ METER selector
Press to turn the LED peak program meters into a tracking meter. Each time the selector is pressed, the function of the meter will change.

⑫ MUTING switch
Normally set this switch to ON.
If the video cassette recorder is not transporting tape at the proper playback speed, or if many dropouts occur due to the mistracking of the video heads of the video cassette recorder, or due to scratches and dust on the magnetic tape, the muting circuit will activate and the reproduced sound will be cut off.
If you do not want the reproduced sound to be cut off by the muting circuit, set the switch to OFF.
See "How to use the MUTING switch" on page 14.

⑬ PEAK HOLD RESET buttons
You can choose either of two ways to have the peak level indicated:
When the AUTO button is pressed, successive peaks are held for about 1.7 seconds, except when a higher peak occurs before 1.7 seconds have passed, in which case that peak is immediately indicated. When the power is first turned on, the AUTO peak indication mode will automatically operate.
When the MANUAL button is pressed, the peak level will be held on the scale until a higher peak occurs, and that peak will be held. To reset the peak held on the meter, just press this button. You will find this method of indicating the peak input useful when you want to know the highest peak of a tape or disc, or when you want to know both the highest peak as well as the intermittent input levels during live recording.

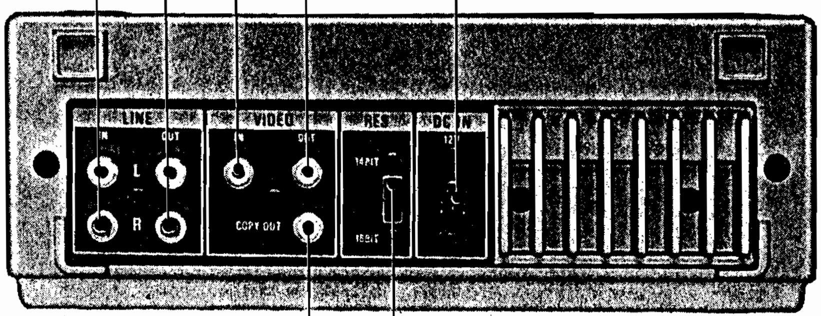
⑭ COPY (digital tape copy) switch
Set this switch to ON for digital-to-digital tape copy, with absolutely no deterioration in signal quality, using a pair of video cassette recorders and the COPY OUT jack at the rear.
Be sure to set this switch to OFF except during digital tape copy.
With this switch set at the ON position, no signal is obtained at the VIDEO OUT jack.
See "DIGITAL TAPE COPY" on page 15.

⑮ INPUT selector
LINE: to record through the LINE IN jacks at the rear.
MIC: to record through the MIC jacks.

⑯ BATT (battery) CHECK button
While this button is kept depressed, the upper meter shows the battery pack condition.

⑰ MIC jacks (phone jacks)
Any low-impedance microphone equipped with a phone plug may be used. If your microphone is equipped with a mini plug, you will need a plug adaptor.

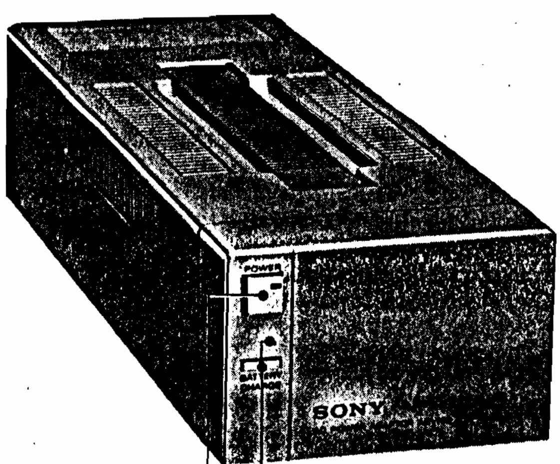
REAR PANEL



COPY OUT (tape copy output) jack
To perform digital-to-digital tape copy, connect this jack with the video input jack of the video cassette recorder for recording so that when the COPY switch is set to ON, playback signals in which errors are corrected and/or concealed are obtained.
Be sure not to use this jack except during digital tape copy. Normal recording and playback cannot be performed using this jack.

RES (resolution) selector
Selects the resolution for recording.
14 BIT: for recording in accordance with the technical specifications of the Electronic Industries Association of Japan (EIAJ) which has adopted the 14-bit linear quantization format.
Set the selector to this position when the tape recorded with this unit is to be played back using another PCM digital audio processor which conforms to the 14-bit quantization format of the EIAJ.
16 BIT: for recording and playing back using this unit with a wider dynamic range and less distortion.
Normally set the selector to this position.
During playback, it is not necessary to select the position of this selector, since the 14-bit or 16-bit format used for recording is automatically selected.
For detailed information, refer to page 25.

AC POWER ADAPTOR AC-700 (SUPPLIED)



SPECIFICATIONS

Signal system	Conforms to EIA television standard, NTSC color
Code format	Conforms to the technical specifications of the EIAJ (standard format using 14-bit quantization), or 16-bit quantization format
Number of audio channels	2 channels
Sampling frequency	44,056 Hz
Quantization	14-bit linear quantizing, or 16-bit linear quantizing
Frequency response	10–20,000 Hz ±0.5 dB
Harmonic distortion	Less than 0.007% (14-bit format) Less than 0.005% (16-bit format)
Dynamic range	More than 86 dB (14-bit format) More than 90 dB (16-bit format)
Channel separation	More than 80 dB
Wow and flutter	Below measurable limit
Error correction	Error correction and concealment using CRCC and parity
Emphasis	Pre-emphasis (in recording): fixed at ON De-emphasis (in playback): automatically switched to ON or OFF (by detecting pre-emphasis identification code) Time-constant: 50 μsec, 15 μsec

Inputs

	Type	Reference input level	Impedance	Minimum input level
MIC	Phone	—	Accepts low impedance microphones.	0.435 mV (−65 dB)
LINE IN	Phono	−10 dB*	40 kilohms	95 mV (−18 dB)
VIDEO IN	Phono	1 Vp-p	75 ohms	—

Outputs

	Type	Reference output level	Load impedance
LINE OUT	Phono	−10 dB**	More than 10 kilohms
VIDEO OUT	Phono	1 Vp-p	75 ohms
COPY OUT	Phono	1 Vp-p	75 ohms
HEADPHONES	Stereo phone	−24 to −48 dB Attenuation: 5 steps (24, 18, 12, 6 and 0 dB)	Accepts low impedance headphones.

* Input level when the peak program meters deflect to −15 dB.

** Output level when the playback level is −15 dB as shown by the peak program meters.

General

Power requirements	Operating voltage: 12 V dc Usable power sources: 12 V dc with the Sony NP-1 rechargeable battery pack (optional) 120 V ac, 60 Hz with the supplied AC-700 ac power adaptor 12 V car battery with the Sony DCC-2400B car battery cord (optional)
Power consumption	17 watts dc
Dimensions	PCM-F1: Approx. 215 × 80 × 305 mm (w/h/d) ($8^1/_2 \times 3^1/_4 \times 12^1/_8$ inches) AC-700: Approx. 107 × 80 × 305 mm (w/h/d) ($4^1/_4 \times 3^1/_4 \times 12^1/_8$ inches) not including projecting parts and controls
Weight	PCM-F1: Approx. 4 kg (8 lbs 13 oz) net AC-700: Approx. 3.2 kg (7 lbs 1 oz) net
Total weight	Approx. 8.1 kg (17 lbs 14 oz) in shipping carton, including PCM-F1 and AC-700
Accessories supplied	AC power adaptor AC-700 (1) Video connecting cord VMC-110C (1) Video connecting cords VMC-1S (2) Audio connecting cords RK-112 (2) Shoulder strap (1)
Optional accessories	Rechargeable battery pack NP-1 Car battery cord DCC-2400B Video monitor cable (with 26-pin multi-connector, two phono plugs and two mini plugs) VMC-220A Plug adaptor (mini plug to phono plug) PC-5A Switch box SB-10 Carrying case LC-170 Carrying handle AH-220

Design and specifications subject to change without notice.

BASIC TERMINOLOGY

A/D and D/A converters
In a PCM system, analog signals are quantized and then recorded as digital signals. In reproduction, the digital signals are converted back into analog signals. Converting the signals into digital signals is performed by the analog-to-digital (A/D) converter, and converting the digital signals into analog signals is performed by the digital-to-analog (D/A) converter.

Analog
The word analog is used in contrast to digital. Analog quantities denote quantities that change continuously like the temperature or voltages. Ordinary audio signals are called analog signals, and VU meters have an "analogy" with the variation in these analog signals which is indicated by the deflection of the pointer. This type of meter can be called an analog display meter.

Bit
This is an abbreviation of binary digit. It is a unit of information equal to one binary decision, so that 1 digit is referred to as 1 bit. Three bits refer to a 3-digit code. With n bits, it is possible to indicate and subdivide 2^n types of information.

Code error
This refers to an erroneous 1 or 0 in the encoded signals. It is caused by dropouts, jitter, noise, etc. If a recording is played back with these code errors, it will come through as a clicking sound. In order to compensate for these errors, a number of methods are used in the circuitry: pre-value holding, linear interpolation and error correction word encoding.

CRCC (Cyclic Redundancy Check Code)
A group of bits or a "word" which detects erroneous data. The probability of detecting erroneous data depends upon the number of bits. In the EIAJ format and the PCM-F1, each TV H (horizontal) line has one "word" of CRCC which comprises 16 bits, so erroneous data are detected at a 99.9985% probability.

Digital
This word originates from "digit" meaning finger and is used in contrast to analog. A digital quantity denotes a quantity by which a variable amount is discontinuously encoded (numerical values). In other words, a digital quantity is an encoded analog quantity. The word also denotes using numbers expressed in digits and in a certain scale of notation to represent all the variables that occur in a problem.

Dropout
This refers to distinct but temporary gaps in the signal level during the playback of recorded data caused by marks or dirt on the surface of the tape. In PCM systems, it results directly in a code error. When burst-formation type of dropouts occur, the errors are scattered and corrected or interpolated.

Dynamic range
In PCM, dynamic range is expressed as the ratio between the maximum acceptable signal input and the quantizing noise. With linear quantizing the range is proportional to the number of bits, but with non-linear quantizing the obtainable dynamic range is much wider with the same number of bits. Thus, in the PCM-F1 which has adopted linear quantization format, the dynamic range of the 16-bit format is wider than that of the 14-bit format.

Encoding
Encoding denotes the conversion of quantized amplitudes into a pulse code. A binary code is most commonly used. In actual operation, the quantizing and encoding are performed simultaneously by the analog-to-digital converter.

Error correction words
In the 14-bit format of the EIAJ, two error correction "words", P and Q, are added to each data block consisting of six "words". Each P and Q corrects one "word" of erroneous data, so that two "words" of erroneous data in a data block can be corrected. With the interleaving process, erroneous data up to 32 H can be corrected.

Interleaving
Interleaving is a method to disperse dropout errors by changing the sequence of information "words" (hereafter referred to simply as word without quotation marks) in recording. Restored to the original order in playback, the erroneous word is invariably placed between the correct words, and thus linear interpolation, etc., can be easily performed.

Jitter
This term denotes the instability of a signal in either its amplitude, its phase, or both. It is generated when signals are played back on a tape recorder with wow and flutter, whereby noise is added to the signals. In PCM systems, it is the cause of code errors along with dropouts in the tape medium.

Linear interpolation
When a word error has been detected, it is corrected by the error correction circuitry. If the error exceeds the error correction capability of the digital audio processor, the average value of the preceding and succeeding words is substituted for the erroneous word. The error is thus interpolated so that there is no audible difference. In the 14-bit format of the EIAJ, burst errors of 32H or more, which cannot be corrected, are interpolated.

PCM
This acronym stands for pulse code modulation. It refers to a system of modulation whereby ordinary signals like audio signals are replaced by pulses, their amplitudes are turned into digital codes, and the resulting signals are transmitted or recorded. All the signals are expressed in binary digits, 1 for every pulse and 0 for every absence of pulse. Therefore, the signals are resistant to noise, and distortion can be kept down to a very low level right up to the high frequencies without being dependent on the frequency.
In the PCM system, the signals are sampled several tens of thousands of times a second and the sampled values are quantized. The code resulted from quantization each time is 42 bits consisting of 14 bits for right (R) channel, 14 bits for left (L) channel and 14 bits for error correction and error detection. The binary code per second is equivalent to a frequency of several megahertz.
A frequency as high as this cannot be recorded by a conventional audio tape recorder. This is why a video cassette recorder, which can record the PCM code delivered by the PCM digital audio processor, must be used with the PCM processor. The PCM processor requires less than half the frequency bandwidth of TV signals. Of course, to be recorded on a video cassette recorder, the PCM code must first be converted into a TV signal.

Quantizing and quantizing noise

Quantizing is the process in which analog signals are translated into a digital code. Quantizing is done by dividing the range of values of the sampled amplitude of an analog signal into a great number of subranges, to each of which is assigned a value. There is a slight error between the original signal and the quantized value. This is heard as noise and is known as quantizing noise, or quantization distortion. This noise is inherent to PCM. The signal-to-noise ratio of the PCM and dynamic range are determined by this noise.

Sampling and holding

This is a circuit which is used in an analog-to-digital converter to measure an analog signal (sampling) and to increase the duration of that signal (holding). With holding, a fixed period of time is required to convert the analog signal into a digital signal.

Sampling and sampling theorem

Sampling refers to the extraction of the amplitude of an analog signal at regular intervals of time.
The sampling theorem (developed by Shannon) states that two samples per cycle will completely characterize a bandlimited signal; that is, the sampling rate must be twice the highest frequency component.

Word

A group of bits that express a single quantizing value is called a word. In the PCM-F1, a word comprises 14 or 16 bits.

SYSTEM AND CIRCUIT DESCRIPTION

Emphasis

The emphasis circuit of this unit is designed to reduce the amount of noise and improve the signal-to-noise ratio by automatically boosting the high-frequency response during recording (pre-emphasis) and detecting the boosted amount and lowering the response during playback (de-emphasis).

LED peak program meters

An incoming analog signal is converted to a digital signal, which is further converted to a video signal and delivered from the VIDEO OUT jack of this unit to a video input jack of a video cassette recorder. When the video cassette recorder is set to the record or record monitoring mode, the video output signal from the recorder is fed to the VIDEO IN jack of this unit. The incoming video signal is converted to a digital signal, then to an analog signal, and the peaks of this analog signal are displayed on the LED peak program meters.
For this reason, if the VIDEO IN jack of this unit is not connected with the video output jack of the video cassette recorder even in recording, you cannot monitor the input signals to be recorded, nor do the meters deflect, though recordings can be made.

14-bit format and 16-bit format

The 14-bit format of the PCM-F1 conforms to the technical specifications of the EIAJ which has adopted the 14-bit linear quantization format. In the PCM-F1, the 16-bit format which is compatible with the 14-bit format of the EIAJ is adopted together with the 14-bit format in order to obtain better results such as wider dynamic range and less distortion. The 14-bit and 16-bit formats can be selected with the RES selector.
A tape recorded using the PCM-F1 with the RES selector set to "16 BIT" can be played back using another PCM digital audio processor which conforms to the 14-bit format of the EIAJ. In this case, the reproduction will be equivalent to that of the 14-bit format. A tape recorded using another PCM digital audio processor which conforms to the 14-bit format of the EIAJ can, of course, be played back using this unit.
During playback, the difference between these two formats is detected automatically.
If a tape recorded in accordance with the 16-bit format is duplicated using the digital-to-digital tape copy function of this unit with the RES selector set to "14 BIT", the format will be converted to the 14-bit format.

Error correction capability of the 14-bit and 16-bit formats

In recording with the 16-bit format of this unit, the error correction word Q of the 14-bit format is replaced by the 15th and 16th bits of the data so that the 16-bit format is compatible with the 14-bit format of the EIAJ. For convenience, we express the word comprising the information of the 15th and 16th bits by a symbol "S" instead of the symbol "Q" of the 14-bit format. In the 14-bit format, data contain error correction words of P and Q; in the 16-bit format, data contain a single error correction word of P. Accordingly, the error correction capability of the 16-bit format is inferior to that of the 14-bit format. If a tape recorded with the 16-bit format is played back by using a PCM digital audio processor which conforms to the 14-bit format of the EIAJ, the error correction capability will be equal to one parity bit of the 16-bit format.

	Usable error correction word	Error correction capability
14-bit format of the EIAJ	Two parity bits of P and Q	Burst errors of up to 32 H can be corrected.
16-bit format of the PCM-F1	A single parity bit of P	Burst errors of up to 16 H can be corrected.

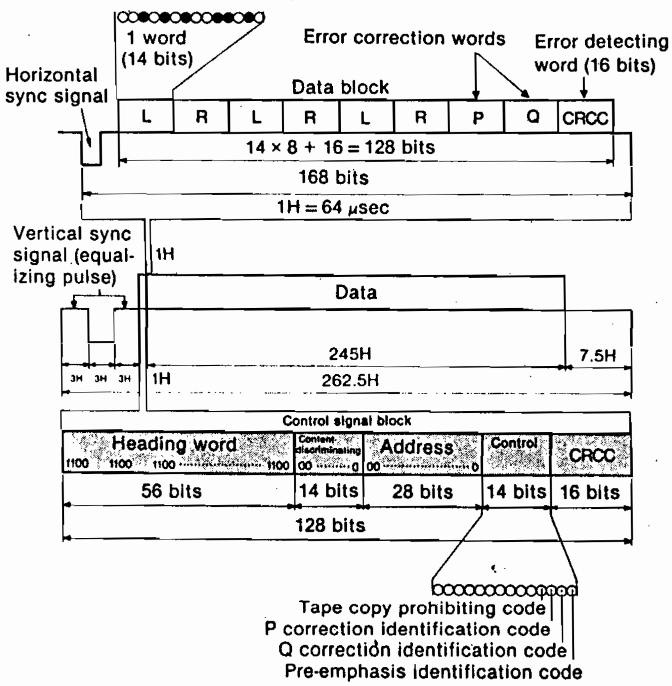
It should be noted that burst errors beyond the error correction capability will be compensated for so that they are not perceptible.

14-bit format of the EIAJ

Interleave array

H 0	L 0	R 48	L 95	R 143	L 190	R 238	P 288	Q 336	CRCC
H 16	L 48	R 0	L 47	R 95	L 142	R 190	P 240	Q 288	CRCC
H 32	L 96	R 48	L 1	R 47	L 94	R 42	P 192	Q 240	CRCC
H 48	L 144	R 96	L 49	R 1	L 46	R 94	P 144	Q 192	CRCC
H 64	L 192	R 144	L 97	R 49	L 2	R 46	P 96	Q 144	CRCC
H 80	L 240	R 192	L 145	R 97	L 50	R 2	P 48	Q 96	CRCC
H 96	L 288	R 240	L 193	R 145	L 98	R 50	P 0	Q 48	CRCC
H 112	L 336	R 288	L 241	R 193	L 146	R 98	P 48	Q 0	CRCC

Signal array and structure within one field



16-bit format of the PCM-F1

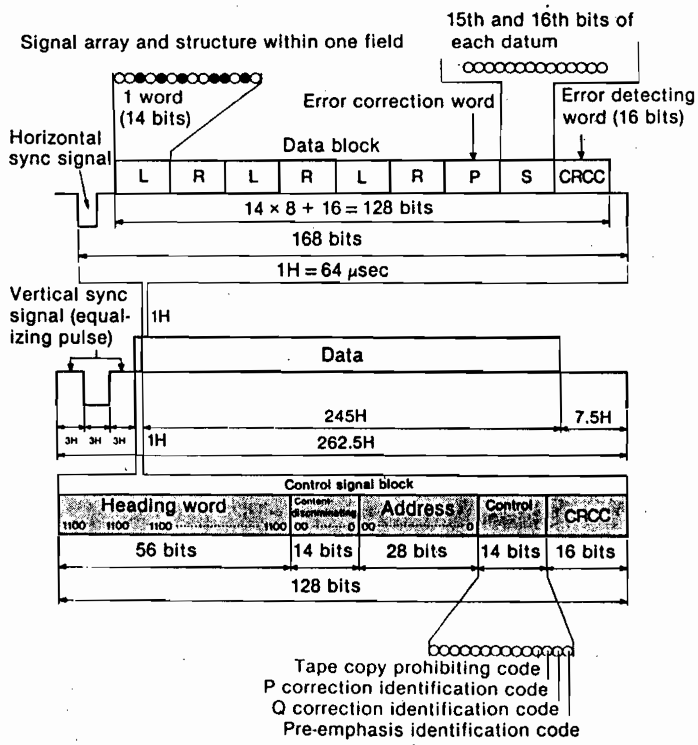
Interleave array

H 0	L 0	R 48	L 95	R 143	L 190	R 238	P 288	S	CRCC
H 16	L 48	R 0	L 47	R 95	L 142	R 190	P 240	S	CRCC
H 32	L 96	R 48	L 1	R 47	L 94	R 42	P 192	S	CRCC
H 48	L 144	R 96	L 49	R 1	L 46	R 94	P 144	S	CRCC
H 64	L 192	R 144	L 97	R 49	L 2	R 46	P 96	S	CRCC
H 80	L 240	R 192	L 145	R 97	L 50	R 2	P 48	S	CRCC
H 96	L 288	R 240	L 193	R 145	L 98	R 50	P 0	S	CRCC
H 112	L 336	R 288	L 241	R 193	L 146	R 98	P 48	S	CRCC

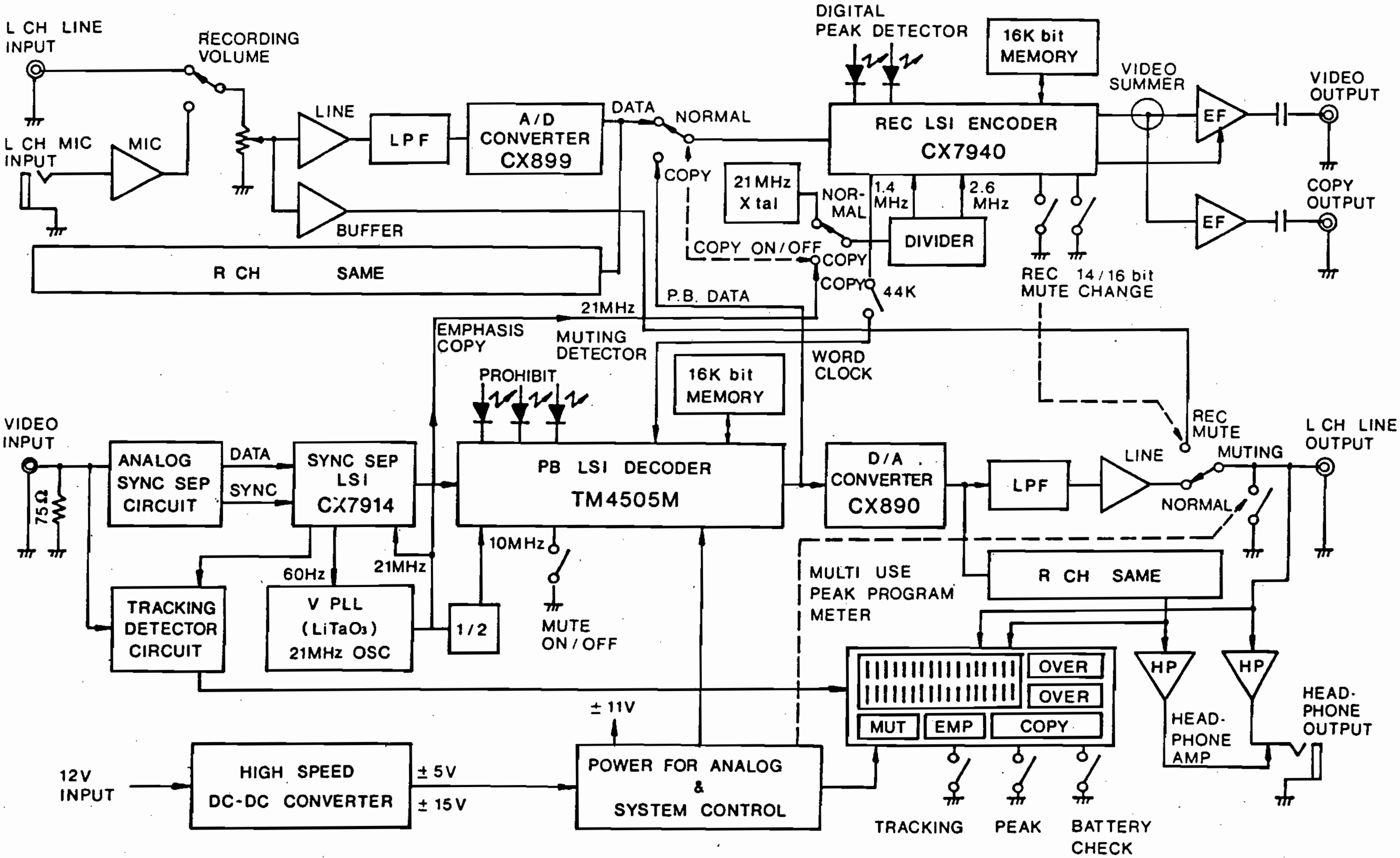
L_{336} R_{288} L_{241} R_{193} L_{146} R_{98} P_{48} ➡ Information of the 15th and 16th bits of each datum

①② ③④ ⑤⑥ ⑦⑧ ⑨⑩ ⑪⑫ ⑬⑭

Signal array and structure within one field



BLOCK DIAGRAM



LIST OF DIAGRAMS