AUDIO TAPE EDITING
In Radio News Broadcasting.

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INTRODUCTION

The tape recorder has been an important tool in news reporting in this country since 1947. It allowed news broadcasters to present the views of newsworthy people, in their own words, without requiring them to be present in the studio at broadcast time. Tape recording thus immeasurably enlarged the scope of radio broadcasting both of news and of opinion. In addition, with the tape recorder being, in effect, a moveable studio, events of all kinds could be broadcast from tape with a minimum of expense and a maximum of fidelity, no longer remaining dependent upon telephone lines sometimes of indifferent quality. Another dividend resulting from the use of tape is the elimination of loss of air time due to technical breakdowns during live air programs. All in all, tape recording has served us well.

The small, light tape recorder is the key to modern reporting. It was not always so! I remember an expedition in 1949 that required four men to get a two- or three-minute interview with a senator on his way back to Washington from New York. The interview was recorded on the rear platform of a railroad car between ~~M~~ Penn Station, New York City and Newark, New Jersey. The recording equipment hauled aboard at New York and unloaded in Newark consisted of a Brush BK401 tape recorder weighing about 50 pounds, two 6 volt automobile storage batteries a motor generator (to convert 12 volts DC to 110 volts AC to power the tape recorder) and spare tape, mike and other equipment. Now the reporter confronted with a similar job need only sling a lightweight recorder no bigger than a camera over his shoulder and be ready for anything, in practically any surroundings. All he really needs is a comprehensive understanding of tape recording and tape editing; I hope the rest of this little book will be instrumental in giving him a good start in that direction.

RECORDING FOR BROADCAST

Part 1. ~~Studio Recording.~~ Fundamentals of Sound

The Nature of Sound

Magnetic recording of sound, as well as ~~any~~ other methods of recording, requires an understanding of the physics of sound. What is sound, actually, and how is it originated and transmitted?

Sound is created by vibration of any material or matter in an elastic medium. One cycle of sound consists of two excursions of sound, one advancing from nothing to its maximum in one direction and falling back to nothing again, the other excursion passing the zero-sound line in the other direction, attaining its maximum NEGATIVE value, then retreating to nothing again. (~~drawing~~). The total number of complete cycles taking place in the space of one second is the frequency of the sound. What we ~~generally~~ call "sound" is confined, mostly for convenience in terms, to the frequencies that lie between 20 and 20,000 cycles per second ~~S~~ (abbreviated "cps"). Frequencies below 20 cps and above 20,000cps are called "subsonic" and "supersonic" sound, respectively. Let no one at this point venture to ~~give voice to~~ REFER TO that ancient chestnut about the tremendous roar in the wasteland where there was no one listening! It is still sound, even if not heard by anyone or anything! Sound waves are as objective as sea waves. Subjectively, of course, there is no sound when we don't hear it; sound is here thought of as the stimulus to hearing. And hearing is an entirely different study, a puzzle that even now has been only partially explained.

THE GENERATION OF SOUND

Sound can be generated in an infinite number of ways, but the final result is always a vibratory motion of some kind that lies within the range of hearing. When you ~~pull~~ pluck a violin string, for example, ~~and let it go~~, you start a sound wave. As the string flies back, it pushes the air in front of it. These particles of air that are pushed by the string, push their neighboring air particles and so on until the force of the string's first push has been exhausted. When that happens, the string is pulled in the opposite direction, both by its elasticity and by the force of the partial vacuum in the space it has just traversed. Another sound vibration then takes place, similar to the first one, differing only in direction (or phase) and strength. The first wave, or "push" wave, is called a wave of compression, since the string caused a compression of the air in front of it. The second wave (or alternation of the cycle) is called a wave of rarefaction, or "pull" wave, since the string was pulled back partly by its own stored or torsional force and partly by the pull of vacuum. This complete cycle of sound will continue to vibrate, at a rate (frequency) determined mainly by the string's various properties, until all the energy imparted to the string by that first "plucking" has been expended in heat.

Do not get the impression, as I did for a long time, that sound waves actually travel in wavy lines~~, like the above drawing~~. They don't! A SINE WAVE ~~The~~ drawing simply indicates, as I said before, a rising PRESSURE ~~travel~~ ~~[illegible]~~ in one direction and then ABSENCE OF PRESSURE IN the other. In sound work, especially, it will help IF YOU ~~to~~ remember that all wave phenomena travel in straighter lines with increasING ~~in~~ frequency. Very short radio waves, for example, shoot straight off into space. The basis of radar is that they do so, then rebound in practically straight lines when they hit an obstacle of any kind. In audio work the same thing is true; shorter waves emanate from the source directly and bounce off hard surfaces, while longer waves tend to follow surfaces and curl around. And remember, sound wave energy is transmitted from one molecule of air, or any other material, to the next - these molecules do not, themselves, actually travel in apace.

Sound is transmitted through any liquid,solid or gas but is not transmitted through a vacuum. The denser the medium, the faster ~~the waves of~~ sound travel$.At room temperature sound travels about 1100 feet per second in air, in water at normal temperatures roughly five times that (5500 ft. per sec.) varying with the water's salinity, and in steel about 15 times the speed in air. Sound is transmitted, you remember, by the energy transmitted from one molecule of matter to the next. A good analogy might be the ~~an~~ educational toy which consists of a number of metal balls hanging ~~connected~~ in a straight line ~~by strings to~~ from a frame. If you pull the first ball back and let it go, it makes all the other balls move in turn, while the last ball in line moves almost as far out as the first ball was pulled back. This oscillatory motion would continue until all the stored force (of the first pull) was expended.

~~If you would like to know how~~ To find ~~out~~ the physical length of a sound wave, you divide the velocity (1100 ft.per sec. in air) by the frequency (the number of complete cycles of that sound per second. Thus, the measured length of a 400 cps. tone would be, for instance, $\frac{1100}{400} = 2.75$ feet

Disregarding electron motion, matter is not normally in a state of motion. Materials are caused to move or vibrate by energy transmitted to them, generally by way of something external to them. (The theory of electron motion tells us that all matter is in a constant state of motion). Blow a whistle, strike a gong, pluck a string or pound a drum ; and sound ensues. ~~We should remember that~~ There are two kinds of sound vibrations: forced and resonant. A forced vibration results when a moving object hits another object. If you struck a tuning fork, then held its bottom firmly against a glass tumbler, the ~~glass~~ tumbler would vibrate at the tuning fork's frequency. This type of vibration is a forced vibration. But if you struck the glass directly with any inert object, a block of wood, for example, the glass would vibrate, after the initial blow ((which would be wide band noise, ~~if you are interested~~)) at its own natural frequency of vibration. However, if you could tune a tuning fork to this natural frequency of the glass, then struck the fork and held it near the glass, the latter would vibrate intensely. This phenomenon is resonance; it is to be avoided like the plague in all audio equipment and processes. In a few words, forced vibration, yes! Resonance, No! An example of poor thinking in this regard caught my attention a few years ago. A manufacturer ~~printed a display~~ advertised ~~advertisement~~ calling attention to his finely built loudspeaker cabinets which utilized rare woods picked for their resonant tones! I phoned him and explained carefully the horrors of sound that such a cabinet promised. He killed the ad. In fact, probably the best speaker enclosure would be made of massive cast concrete.

Transient vibrations versus sound transients.

The above are two terms that must be understood for a good comprehension of sound. When a material is forced to vibrate because of the sudden application of an external force it begins to vibrate violently at first. This initial vibration may range momentarily over the entire audible spectrum and produces what we classify as noise. The time consumed in this noise producing effort is the transient period of the material. We design equipment to have as short a transient period as possible, as the next paragraph points out. However, there ~~is~~ are contained in musical sound very short elements called sound transients. Good equipment should be able to record these sound transients faithfully.

Damming out of Transient vibrations.

The motion of any element of a transducer [a transducer, as the name implies, is a ~~piece of equipment~~ mechanism that converts SOUND energy from one form to another, e.g. microphone, loudspeaker, ~~photoelectric cell~~, magnetic ~~tape~~ RECORD or reproduce head, etc.]. that is to have as short a transient period as possible must be damped by some means. Microphones and loudspeakers utilize air chambers of various kinds to maintain steady, forced vibrations. Viscous fluids, elastic solids and other materials are used to dampen motion, ~~to~~ thereby reducing transient periods. Electrical circuits may be damped by "loading" with resistances. Thus, any sound, to be transmitted faithfully from origin to end product must encounter a minimum of both resonance and transient periods in its travels

Distortion of Sound

~~1. Phase Distortion.~~

I suppose the first thing to learn about sound is that is that there is no practical way to produce sound, to amplify it, or to hear it, without distortion of many kinds. Sound is distorted in air, in amplifiers, transducers, in our hearing system and probably in our brains, although no one yet has proved the latter. I shall outline a few of the common distortions so that you may know how they occur and may be able to do something practical towards ~~eliminating or~~ reducing them.

I. Phase Distortion.

Two waves of sound ~~at~~ of exactly the same frequency will add together and produce the same sound at higher intensity, if they are are precisly in step with each other in time. That means that ~~the~~ two compression and the two rarefaction waves coincide exactly. If these two waves, also of the same frequency, but slightly out of phase (one has a strong compression wave ALMOST coinciding with the other's weak rarefaction wave) there will be heard a weak version of the sound. The opposite phase of one wave will detract from the intensity of the stronger compression wave. If these two waves should be exactly out of phase, you would not hear any sound at that frequency. There are many variations on the above theme. The most common one is the production of beat frequencies, which can happen when two almost similar sounds are produced at the same instant. The result is alternate degrees of aiding and cancellation of sound, making it appear that the sound was varying in intensity.(loudness). Competent broadcast technicians will always make sure before using micophones that they are all in phase. The way to test is to ~~line up~~ group the microphones and talk into two at a time. If the VU meter shows that two mikes produce less voltage than one alone the mikes are out of phase and the connections of one of them must be reversed. Also, in using many microphones at a time in one studio, they should be tested in ~~locations~~ groups where one will aid the others; out of phase POSITIONING of mikes can occur, also.

Reflections of Sound

Sound is reflected from hard surfaces of walls, ceilings and floors in much the same way as light is reflected from mirrors. Soft surfaces like draperies, acoustical tiles and such absorbent materials, ~~sop up~~ absorb sound, more at high ~~wavelengths~~ frequencies than low. Thus, in a "hard" or reflective room, a sustained sound may periodically cancel itself as reflected waves of the original sound first cancel (opposite phase) and then aid (in phase) the original wave. When a sustained wave of this pattern occurs (standing wave) and the microphone is located in the cancelled area, very little of that particular sound will be picked up. The opposite also happens; if the microphone is located where the maximum of an aiding phase occurs, a disproportianate volume of sound at that frequency will be picked up.("WOLF" tone).

~~Xiffractionx~~

Dissipation of Sound, and Echoes

In free space (a condition that practically never exists!) sound dies away as its energy becomes dispersed. This dissipation of sound takes place in inverse ratio to the square of the distance from the source. Thus, if we stand ten feet away from a sound source a certain amount of its energy will reach our ears. Twenty feet away one-quarter of its energy, and at 30 feet one-ninth will be heard. Sound energy ~~you know,~~ is used up in overcoming the inertia of air, or other, particles.

In all space, except in a vacuum, (or anechoic chambers) sound is reflected by natural or artificial means. Echoes occur in nature when sound is reflected from various bodies or surfaces, such as hillsides, clouds, buildings etc. Multiple echoes, such as thunder, occur when sound is reflected several discrete times from various surfaces.

Reverberation.

Reverberation is a series of echoes taking place in such rapid succession that they overlap each other. Reverberation in rooms and auditoriums can be reduced by various acoustical treatments such as draperies, tapestries, rugs, absorbent felt, acoustic tile, or even specially constructed and lined absorbent boxes.

Reverberation time is the time it takes, after a sound has been produced, for it to die away completely. For recording speech with clarity, reverberation time should not be longer than half a second. Longer reverberation times may enhance the sound of organ music but tend to detract from the enjoyment of complex music, i.e. symphony orchestra music and chamber music. Nothing longer than three seconds can be recommended, even for organ. ~~It is extremely disconcerting to try to understand speech in an extremely reverberent hall!~~

Audiences can be quite a large factor in "damping" reverberation, especially when they are wearing damp clothes. (Forgive the possible meanings of this unfortunate wording!) In humid atmospheres you must expect to find less than normal reverberation. More than normal reverberation (of short time) will make sound appear brighter or "live". Less would make sound appear duller or "dead". I refer to quantity of reverberation here, not time of reverberation! ~~Longer Reverberation time than half a second tends to make~~ ~~Xxxxxxxxxx~~ speech unintelligible.

Microphones.

Any number of microphones that feed into a single amplifier constitute, electrically, a single ear. Such a hook-up is called, naturally, monaural. Two microphones mounted on a replica of a human head, with microphones instead of ears, and connected separately to two amplifiers, etc., to two separate headphones is a two-eared system or binaural. The ~~latter~~ binaural system, incidentally, gives the best approximation of being actually there on the spot, if one excludes possible phase shift ~~due to the amplifiers~~ in the circuits. Stereophonic systems must have at least two microphones, placed a distance from one another, feeding two systems ending in two or more loudspeakers installed the same distance from each other as the microphones were. Of course if this distance was not known, you could approximate it by cut-and-try methods of speaker placement. Microphones are transducers that convert sound energy into electrical energy and are subject to all the laws governing the transmission of sound. They should not produce peaks at any audible frequency and should be critically damped so that they can translate sound into electromagnetic waves with great fidelity.

There are quite a few types of microphones:- condenser, ribbon, dynamic, crystal and ~~over-ceramic~~. However, you need not, for your purposes, study each type. You need to know only how they differ from each other. Condenser microphones of the highest grade are used for studio music pickup mainly because they give, with proper handling, the best results. They pick up sound, even from a distance, with great fidelity. Dynamic microphones are sturdiest and are best for field use. Ribbon mikes are difficult to use, must be phased correctly if more than one is used, and distort a great deal if used too close to the mouth OR INSTRUMENT ~~on speech~~. Here are a few general admonitions on the handling and use of microphones:

1. Don't abuse mikes! Don't tap them, blow into them or overload them. They are most delicate in construction and can easily be overloaded, causing considerable distortion.
2. Use very low impedance mikes wherever possible. This allows long cables (100 feet or more) to the tape recorder without hum pickup or much loss ~~diminution~~ of level of sound. By low impedance I mean from 50 to 200 ohms.
3. If you HAVE to use a high impedance mike, and HAVE to use a long cable, you must do either ~~one~~ of two things. You can use a stepdown transformer to make the cable effectively low impedance and feed it into a low impedance hook-up of ~~into~~ your tape recorder. Or, you can use an extra amplifier, feed it with your mike, and feed the cable from a low impedance output ON ~~from~~ the amplifier to the proper input of your tape recorder. What I am emphasizing is only that you cannot use a high impedance mike more than 10 feet or so from your tape recorder without picking up hum.
4. For all round news work I think it best to use low impedance dynamic microphones. One mike can be held so that it favors the speaker, it will not pick up too much distant sound and, at low impedance, will not be too sensitive to handling. If required, a dynamic mike can be put on a stand with the head, or diaphragm, pointing up. In this position it is omnidirectional and can be used for a close-in group interview if necessary.
5. To reduce wind noise, use any handy porous material wound around the mike. ~~xxx~~ The idea is to reduce the wind velocity at the diaphragm of the mike. I have done so by almost covering the mike with ordinary black friction tape. Another way is to static ~~tip~~ a small rubber balloon ~~xxx~~ over the mike; most of the wind will be shunted away while direct sound will actuate the mike.

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Clean the heads and every point the tape touches either with alcohol or a cleaner designed for the purpose.

Recording the News.

Long before you go out to record a ~~news~~ program of any kind, the tape recorder must be ~~checked~~ to make sure it is in good operating condition. It should be loaded with fresh or well-erased tape and a test recording made. If there is a recording level meter on the recorder, observe the levels at which the best recording takes place. Don't overload the tape with sound, thus causing distortion. Listen to a playback of your test recording to make sure the quality is top grade. If the recorder operates on battery power, see that the battery cells are full power; there is usually a meter to test their condition. After becoming accustomed to the machine you are using you should be able to tell when the levels are right without listening to the monitoring headphomes, but, until then, listen to the headphones while recording, not to your interviewee directly.

Before you record an interview on location, record a few minutes (2 or 3) of the sounds you hear around you. Also, if your interviewee is nervous, record a few minutes of his talk (and yours) that can be thrown away if not needed. The worth of these preliminary recordings may be demonstrated when you edit the spot. Very often a word, or sound, from these preliminaries might come in handy to perfect the editing. If you are obliged to record where there is a constant noise, find some way to explain your noisy surrounding. It is, also, better to fade in (turn on the record level from zero to normal) slowly to background ~~xxx~~ sound at the beginnin and fade out at the end when there is noise of any kind. This fade material will make it possible, if necessary, to cross fade from one segment to another in your edited program material. The fades should be about five seconds in length.

It seems to be customary nowadays to record on miniature recorders. Only the most expensive have the facility to record at the higher speeds which make tape editing easier. Thus, if your recorder is a cassette recorder or low-speed reel-to-reel recorder, it will be necessary to re-record before editing. In order to avoid more than one re-recording session, it is best to re-record all material from cassettes or low-speed reels onto a master editing reel, recording at 15 inches per second if possible, at the same time standardizing the levels and making any needed corrections in the sound quality. For example, a tape ~~containing~~ WITH A heavy hum could be routed through a filter to remove as much of the hum as possible while re-recording. Or a tape of an indistinct speaker could be filtered to emphasize higher frequencies and make his speech clearer. In some cases a slight amount of artificial reverberation will do the same thing.

At the same time that re-recording takes place it might expedite matters if the news director listened to the material for quality and content, ~~xxxxxxxxxxx~~ and timed the several pieces accurately. If possible, an accurate script should be typed as an aid to editing.

TAPE EDITING

This expl~~ana~~ication of tape editing will ~~consider~~ deal with only ~~that of~~ quarter-inch full track audio tape, although any audio tape up to two inches wide, with any number of tracks up to ~~and including~~ 24 tracks can be edited in the same way, the only caveat being that all the tracks must run in the same direction.

Magnetic tape has , ~~almost~~ since the beginning, been the preferred medium for original sound recordings, simply because it is so easy to edit. By editing I mean actually cutting and splicing the tape, not ~~the~~ "re-record editing" which had previously been the only way to "edit " disc recordings. Note again that only recorded tapes running in the same direction can be ~~physically~~ edited this way. ~~– none other~~!

The common tools for ¼" tape editing ~~is~~ are the Editall Block,(which I invented in 1947, patented in 1952 and have manufactured ever since) a single edge razor blade, ~~generally~~ a black or yellow grease pencil and, most important, an education in the art of tape editing. For it is properly an art in its highest aspects, which include a good understanding of speech, hearing, acoustics, phonetics and ~~[illegible]~~ other barely related technologies.

Block face for photo 2"

How to use an Editall Block   Drawing of block!

The function of the block is to hold ~~the~~ tape firmly, without slipp~~age~~ING, while it is being cut and spliced. ~~Present-day~~ Tape is coated on one side only and the coated side should always be face down in the block for splicing, no matter on which side it is cut. (All Editall Blocks are designed to be ~~firmly~~ PERMANENTLY mounted in a convenient spot on the recorder, with the slanted cut to the left). When marking the tape for editing with the grease pencil (this marking should be eliminated when you become more expert) the marking is done on the BASE side of the tape, generally glossier but not always! Then, when ready to cut the marked tape, place it in the groove of the block with the base(marked)side f facing up. The tape is forced into the groove by finger pressure and conforms to the curved bottom of the groove. Tape slightly narrower than standard will not touch the bottom but will still be held ~~fairly~~ firmly. Tape wider than standard, up to .250", will still get into the groove but will require more force to slide it in the groove.

(Standard width for ¼" tape is .246"-.248").

Audio tape should always be cut AT the 45-degree cutting slot, using a single-edge razor blade with its corner inserted into this slot before slicing the tape. After the edited-out section of tape has been discarded, butt the remaining ends together firmly and lay a length of splicing tape 1 inch to 1½" long evenly over the cut edges. (Splicing tape for use with quarter-inch tape is commonly 7/32" wide. I helped set this standard in the late 1940's). Use Mylar(polyester)base splicing tape if possible. It lasts longer and gives less trouble. If you want the best possible splices, you should buy Editabs, a prepared splice I designed in 1965 that practical makes it impossible to make anthing but a perfect splice. Whatever kind of splice yo use, smooth it down firmly with a fingernail. To remove the spliced tape from the Editall Block, grasp both ends of the spliced tape at both ends of the block, pull the tape taut, then lift it up out of the groove. DON'T EVER ATTEMPT TO PEEL THE TAPE out of the groove! If you do, you will ~~just~~ destroy the tape's useful life completely by ruffling the tape edges. After practising the above procedure on scap tape a few times you should be able to make a good splice in less than ten seconds. An expert can do it ~~all day~~ in less than three seconds.

Make sure, before you begin to edit, that your razor blade in not magnetized. If you cut tape with a magnetized blade, each edge of the taper will be in turn magnetized and will cause a tiny click each time it passes the playback head. Either get a fresh unmagnetized blade or demagnetize it with a degausser or a small head-demagnetizer.

Establishing a Tape-Marking Point. See 10 A

The place where what is recorded on the tape is converted into sound is the magnetic gap of the playback head. That is where logically the tape should be marked for cutting when you edit. However, the playback head is often completely covered by a magnetic shield or is otherwise difficult to get at. The solution is to establish a tape-marking point somewhere to the right of the playback head, within a space of two to three inches, where you can mark the tape. After finding such a point, measure its distance from the center of the playback head and duplicate the same space on your Editall Block. ~~Thereafter, you just mark the tape at this tape marking point~~

Also, It is best to avoid marking the tape on the playback head for several reasons. One is that repeated pressure on the head may push it out of XXXX proper alignment and require delicate readjustments. Another is that grease pencil marks are abrasive and can cause uneven wear of the polished surface of the magnetic head. Still another reason is that the slightest amount of any kind of material on the head surface can reduce appreciably its ability to reproduce high frequency sound, since it does not permit the tape surface to conform to the head surface as closely as it otherwise would.

Establishing a Tape-Marking Point.

The place where what is recorded on the tape is converted into sound is the magnetic gap of the playback head. Logically, that is where the tape should be marked for editing. However, there are several reasons why this otherwise logical spot should not be used. 1. The playback head is often covered by a magnetic shield or is otherwise hard to get at. ~~Also, it is best not to mark tape on the playback head for these reasons~~: 2. Repeated pressure on the head may knock it out of alignment, requiring delicate readjustment. 3. Grease pencil marks are abrasive and can cause uneven wear of the polished surface of the magnetic head. 4. the slightest accumulation of pencil marks on the head surface can reduce appreciably the high frequency response of the tape recorder, since, then, the tape surface cannot conform closely to the head surface.

Probably the simplest way to establish A tape marking point AWAY FROM THE HEAD is as follows: Record a few seconds of any continuous sound on tape. Cut the tape perpendicularly at any point WITHIN THIS SOUND, AND splice onto it a few feet of paper leader tape. Pull the recorded tape through the tape recorder until you reach the spot where the sound stops. That will be where the recorded tape ends and the leader tape begins. Find a place where you can mark the recording tape within two or three inches to the right of the playback head. Mark the tape. Then take it out of the recorder, lay the tape in the Editall Block with the joint between the recorded tape and the paper tape bisecting the slanting cutting slot in the block. Mark the block permanently wherever the tape mark falls. Thereafter all you have to do is to mark the tape at the point you have selected, put the tape in the block, line up your mark on the tape with the permanent mark on your block and cut at the slant cut. If you have done the job precisely your editing will be as accurate as it is possible to make it. When you have practised enough in this method, you will be able to eliminate the tape-marking step. That is, you will THEN grasp the tape between thumb and forefinger at the tape-marking point and lay the tape down in the Editall with the fingers lining up with the permanent mark. That will give you a fairly accurate location and make splicing still faster. Of course, if top accuracy is needed, marking the tape is still best.

Incidentally, when tape is marked for editing in this way we get an additional dividend. Sometimes, while editing, we may cut out a word or a phrase as a trial edit, while we check the sequence for naturalness. Then we may dedide we do not like the result and that the cut-out piece ought to be put back in context. Now, tape has no built in direction indicator; the only way we can ordinarily find out the direction of recording is to put the piece back and play it. But, if the tape has been marked with a grease pencil, we know that the pencil mark will be on the trailing end of the tape and we can be sure of direction before splicing it back in to context.

The Art of Tape Editing.

The kind of tape editing we are ~~going to talk about~~ CONCERNED WITH means cutting from one sound to another so expertly that the transition goes unrecognized. ~~In order~~ To be able to edit tape like this requires a comprehensive knowledge of every factor ~~concerned~~ in the creation of sound: speech, language formation, phonetics, and music. It also requires a fairly good understanding of acoustics, hearing and the human brain. I do not expect that anyone reading the next few pages will automatically acquire ~~all~~ the above ~~noted~~ knowledge. But a thorough study of this treatise, plus a great deal of practise, will, ~~I am sure,~~ provide a good foundation in the art of tape editing.

Recognition of sounds at low speeds.

Sound on tape is made up of a series of magnetic marks. Let us say that these marks have been recorded at a speed of 7½ inches per second (7½ ips). When we play them back at that speed, the reproduced sound is exactly like the original, depending, of course, on the fidelity of the particular recorder used. If sound recorded at one speed is played back at a lower speed, the frequency decreases and ~~the~~ constituent sounds are "dragged out". In this way, if a tone of 1000 cycles per second (cps) were recorded at a speed of 30 ips and played back at a speed of 15 ips, the frequency of the tone would be 500 cps: if you halve the speed you reduce the frequency to half its former frequency. And so on down the scale. At 7½ ips the tone would be 250 cps and at 3¾ ips the tone would be 125 cps. It is obvious, then, that if we are to recognize sounds at the speed with which we pull tape through a recorder by hand while editing, we will need considerable practise in recognition of sound at low speeds. Do not lose sight of the fact that you have to recognize the sound before you can decide where to cut the tape. Practise in learning to recognise sounds could begin with learning the difference between the sounds ~~like~~ of f,v,s and z, and t,b,p,d and so on. For example, at very low speed t and s sound almost alike; the beginning tongue click of t is almost the only difference in the sounds. After you have learned to recognize all the consonant sounds, learn the vowel and diphthong sounds - those of a,e,i,o,u and their various combination sounds. Spend time on learning ~~the~~ to differentiate between almost similar sounds. It would help ~~you~~ a great deal if you learned, at this time, how to achieve a low but constant speed in turning the tape reels by hand.

~~Listening in reverse.~~

Listening in Reverse

In trying to separate almost similar sounds on tape at low speeds it sometimes helps to listen in reverse. When we listen to sounds in reverse we cannot recognize language as such and are forced to listen to minute variations in sound.

This reverse-listening technique is especially valuable when trying to cut within a word where one sound slurs into another in the normal listening mode, e.g. IN a word like "room" where you want to eliminate the "r" sound. There is one thing to watch out for in reverse-listening editing, however: there are some speakers who never enunciate parts of words. Although the words sound complete in normal listening, when we listen in reverse we find they are not! Thus, in editing this speaker we are forced to leave space for those unvoiced parts of words. This phenomenon indicates that we supply parts of words out of our imaginations when they are not voiced.

An example of the use of reverse-listening in editing follows: A reporter crossed from England to France aboard a channel steamer. As it happened, he called the steamer a "streamer", making it necessary to eliminate the "r". Cutting out the "r" in "streamer" at 7.5 ips is not easy editing. The "t" sound is short and not very pronounced, the "r" is short in duration and not easily recognized at low speed. By listening in reverse, however, it was possible to isolate the "r" and cut it out. In practising reverse editing you will notice that beginning hard sounds like "t" "k" and so on are more softly voiced, naturally, than in the normal direction of playback. The tongue clicks that normally precede are then following. Good, clean reproduction and a quiet editing room are required if you are to hear these sounds clearly.

Speech Characteristics in Editing.

After you have learned how to edit speech sounds of all kinds and combinations, you are ready to undertake cutting connected speech. In this kind of tape editing you will notice that ~~each~~ a speaker can be identified by his own characteristic ways of speaking. Since the basic requirement of good tape editing is to produce an edited version that in every way appears original, you must include in the edited version all characteristica like mood, pace, speech habits, and inflections.

Mood.

One of the most fascinating aspects of speech editing is that we are dealing with human sensibilities. We learn to recognize the mood of a series of words. We sense the speaker's emotions in the way words are spoken. However, it is easily possible to ruin a fine expression by careless editing: edited mood must match original mood. In condensing a speech to conserve time, for example, one must not jump from part of a sentence in one mood to part of another in another mood. Even if the edit makes perfect sense on paper it will be nonsense if edited that way.

It will immediately make it apparent that the sentence has been edited. Few people remain in any one mood for any length of time, always excepting, of course, the politician who is reading a ghost-written speech he has never seen before: his mood will be simple caution and nothing else! Moods change, though ever so slightly, and the good tape editor will take notice and do his best to accommodate them.

Pace.

Every speaker has his own pace in speaking; it will change according to his mood, but even then it will be his own pace and not subject to change by the editor. You must realize that pace ~~concerns~~ INCLUDES the ~~periods~~ SPACES between words as well as the time taken to speak a certain number of words. We must allow for breathing space, whether we actually hear the breaths or not. I have always, wherever they are not too time consuming, left in place the "ers", "ahems" and most of the other vocal tricks that people use when speaking. They ~~identify~~ PERSONIFY the speaker, as well as telling us what kind of speaker he is; don't cut them out unless easily done without destroying the pace of speaking. There is a way to maintain a speaker's pace that is also an efficient way to edit speech: before cutting, say the edited version to yourself. If you can say it easily and in good rhythm, you can cut the tape to sound the same way. Then, by all means, cut from sound to sound. Never cut in the middle of so-called "quiet" tape unless it is unavoidable. For example, suppose we had to edit the sentence: " John, my big brother, is home in bed". We want to cut out "my big brother". How would you say what remains? Would you say: "John"--pause--"is home in bed", or would you say, "Johnis home in bed"? If the word "John" was accented correctly, the normal edit would leave practically no space between "John" and "is", "Johnis". In case "John" wwas accented too much, thus indicating that in the original a consonant began the next word (my) it would make a smoother edit to cut out the "i" of "is", making the finished sentence "John's home in bed". The final criterion of all editing is how the finished piece sounds to a new listener. If it sounds natural, it is right: if it does not sound perfectly natural, it is completely wrong!

Speech Habits

There are certain speech habits that can perplex even the experienced tape editor. Some speakers have accustomed themselves to clipping parts of words, especially at their endings; others slur over some sounds and do not even pronounce others. In editing speech by these types, you must leave space for those unvoiced sounds (or, more accurately, space where these sounds would be if they had been voiced!) Again, for emphasis, I repeat that speech habits, even though bad habits, must be retained in editing, or the character of the piece changes. Strive for naturalness at all times. Mechanical speech, perfect as it may be, is not human, not natural and not credible.

Treatment of Rhythmic sounds.

When there is a regularly reoccurring or rhythmic sound in the background of the speech you are editing, you must edit to the rhythm of the sound, or your version will not sound right. In some cases this can be done without too much trouble, even if it entails transposing rhythmic sounds in the clear from one spot to the other in the edited tape. In other cases, the only way it can be done properly is to rerecord (dub) the repetitively patterned sounds into the background. This operation requires the use of three machines, one with speech, the other with rhythmic sounds, both simultaneously recorded on the third machine IN SYNCHRONISM. It is well to observe, in regard to background sound of any kind, that although the listener will disregard it when it is there, he will be shocked if it suddenly disappears and returns, unless there is a natural explanation in the sound itself.

Inflection.

Inflection in pronouncing gives words definite meanings, often at variance with the meanings of the words as read. [In fact, (to wander for a moment from my text) I have advocated for many years that court proceedings be recorded upon an edit-proof medium so that inflections of words be available to higher courts so that prejudice due to inflection can be judged].

In ordinary editing, it is the way in which speech is inflected that makes speech musical and interesting. Read a speech and then listen to a recording of the same speech. A wealth of meaning is added simply by intonation and inflection. Inflection poses an editing problem that is practically insurmountable. It is difficult to use a heavily inflected word except in its normal context. The only exceptions are: 1. A speech may be cut right after a heavily inflected word, any sound like a cough, a clap of applause or another natural interruption inserted, and then continue as desired with anything that will make sense. The interruption causes listeners to forget the inflection. 2. Follow a heavily inflected word with any other word that begins with the same sound as that which originally followed it.

An example of the use of rule #1: We want to edit "On Wednesday next the ship will dock" to read "On Wednesday the ship will dock". The inflection of the "day" of "Wednesday" is upward. If we cut out "next", insert a cough, then proceed with "the" the edit is natural and understandable to the listener. Rule 2: We cut "next" and, if we can find the word in the right background, substitute the word "noon", if that makes literal sense. The reasoning for this rule is that people shape their voicing ~~components~~ MECHANISMS in anticipation of the next word to be spoken. To repeat: people shape their lips, tongues and other voicing mechanisms in anticipation of the sounds to be voiced. The more you study speech editing the more you will appreciate this fact. The safest way, then, to edit speech complicated by inflection, as almost all speech is, is to avoid cutting after an inflected word unless you can use either of the two rules I have noted above.

Cutting within sound

Normally we cut from the beginning sound of the word we plan to cut out to the beginning sound of the word we plan to retain. For example, in the sentence "Editing according to the rules we are following is not difficult" the obvious way to eliminate ~~this~~ the phrase "according to the rules we are following" would be to cut from just before "according" to just before "is". There is another way to edit this sentence, a more professional way. That is to cut in the middle of the "ng" sound in "editing" to the exact same "ng" sound in "following". This is the way that fine music editing is done and professional editing of speech wherever the opportunity presents itself. Cutting within sound makes the edit perfectly natural and noiseless. It is of great value where a speaker mispronounces a word and corrects himself abruptly, generally in the process giving the corrected word unnatural emphasis. For example, a reporter says," The President returned to Washling-- WASHington by train". The "within sound" method of editing would have us cut from the middle of the "sh" sound in the mispronounced "Washling" to the middle of the same sound in "WASHington". The result is a perfectly natural "Washington" with normal accent. Where ever ~~it is~~ possible ~~therefore~~, it is ~~always~~ better to edit this way.

Correction of "level" (intensity of sound).

Those factors of speech like mood, pace, inflection and level are ~~generally~~ present at the same time in the same phrase or sentence. Often , when a speaker accents a phrase, the intensity of sound that his voice projects increases. Or he may drop his voice level while changing his pace. These changes in level, although natural in the original speech, may create problems in editing to shorten the speech, which is the normal reason for editing in radio broadcasting.(In cidentally, I should mention here that in my ~~21 years of experience in radio broadcasting, and my~~ years as a tape editor from 1947 to 1963, I have never been asked to change the meaning of a news piece by editing. The sole reason for editing ~~in the great majority of cases~~ is to shorten a piece while retaining its meaning. (In a small number of cases editing was called for in order to correct pronunciations of words by people in the public eye.). Even if, as I have suggested before, you have rerecorded before editing to corredt levels, it may not have been possible to reduce a sharply increased accent to normal. For example, A speaker corrects a stumble in the sentence: "The train stopped at Balt--WASHington at three o'clock." If the "Balt-" were cut out, the complete correction would sound unnatural. The technique for correction to normal level is as follows: Edit out the "Balt" as usual.

Now play the tape by pulling it through at low speed, marking off, with a grease pencil, the beginning and end of the accented "WASH". Then cut a tiny sliver of splicing tape as long as the "WASH" sound and possibly one third the width of the magnetic tape, and affix it to the oxide side of the tape between the two grease pencil marks. What you are doing is to separate this short piece of tape from the playback head by a few thousandths of an inch, thus reducing its level considerably. The effect of separation of tape from head is to reduce more of high frequencies than low, but the result should be, at least, far better than an unexplainably accented syllable. The amount of level reduction can be manipulated by making this sliver of splicing tape narrower, for less reduction, or wider, for more. After airing of this ~~/~~ tape, do not forget to remove this sliver if the tape is to be erased for reuse. It would ruin a subsequent new recording if allowed to remain in place!

Background Sound

We have now reviewed the bare fundamentals of speech editing. There were no complicating factors involved except those encountered in normal speech - pace, mood, inflection and level. We have taken ~~xxx~~, in sound, what photographers would call a "flat" photograph - our sound has only one dimension. Now we are ready to add another dimension, that of background sound and to learn how to edit tape complicated by changes in background sound. Very little is recorded in a uniformly quiet background in the field/; some characteristic sound that tends to "place" the recording is generally present. During recording, of course, the recordist ~~remember~~ should try to control background sound in the only way he can - by finding a place in which to record where rhythmic sounds (as previously discussed) and disturbing background sounds are absent in whole or, at least, in part. But it seldom happens, even in studios, that background sound can be completely eliminated. Even studios ~~constructed~~ designed by acoustic experts have their own characteristic tones which identify them to these aware of acoustical differences. In the field, traffic noises, a factory whistle, ticking of a clock - a multitude of sounds - may be recorded in the background while an interview is recorded in the foreground,. Now, although background sound is inevitable and enriches the overall pattern of sound, it does complicate editing. The ways in which we must edit ~~against~~ within background sound entails a study of some of the principles of hearing, because, if we can understand how we hear, we can find out how to make changes in background less ~~un~~noticeable. ~~to the hearer.~~

How to use limitations of hearing in editing.

~~The~~ Human ~~ear~~ hearing is extremely sensitive, even though our sense of hearing is not as efficient as that of some animals. Most of us never appreciate our hearing sensitivity, living as we do in noisy environments. We hear best those sounds whose frequency is about 3,000 cps. At that point we can almost hear, in the absence of noise, the sounds created by the collisions of particles of air in space. Our hearing decreases in sensitivity on both sides of 3,000 cps; down to the normal lower edge of our hearing, at about 20 cps, to the approximate higher limit at 22,000 cps. As we age our hearing ability generally decreases because of ~~many~~ various debilitating factors and diseases. This discussion will be confined ~~itself~~ to normal hearing.

Hearing Fatigue.

In editing recordings made within changing backgrounds ~~sounds and noises~~, we have to make sure that we can hear ~~always~~ clearly everything recorded. Now, it seems to be a peculiarity of human beings (possibly animals also, but I cannot vouch for them!) that we sense things more accurately when we do not try too hard to do so. We see best, think best and see best, etc., when we do not strain for perfection. The hearing system tires quickly during a long session ~~spell~~ of editing. Once it has become affected by ~~this~~ "hearing fatigue" our then defective hearing can cause us to make errors we would not have made under normal conditions. ~~of hearing.~~ There are only two ways we can partially avoid incapacitating hearing fatigue: 1. Stop editing and rest frequently: 2. Alter the reproduction curve of your amplifier equipment to make listening as easy on your hearing as possible.

A few pages back we described methods of learning recognition of sound at low speeds when moving tape manually. As you realize, this necessitated learning to recognize sounds heard at very low frequencies. Should both your loudspeaker and your hearing be inefficient, comparatively, at low frequencies, you would either have to strain to hear, or else turn up the volume considerably. Either way, you would become rapidly fatigued. Besides, when the intensity of sound is raised almost to the point of pain (to your ears) your pitch recognition changes. One half of the cure, then, is to use a low-boost circuit in the playback monitor amplifier. That will enable you, ~~I hope,~~ to tell the difference between what sounds like two similar grunting noises! Now, we still have to hear high frequency sounds clearly at modest sound levels in order to be able to tell the difference between a "t", "s" or "d" sound at very low speeds. That requires a high frequency boost. Thus, if the acoustics of the editing room are fairly good, we get sound that is higher at both ends of the frequency curve than in the middle. (We really don't need any boosting between 250 cps and 5,000 cps). It is best to have the boost controls variable so that you can edit at the sound intensity at which you find you function best for the longest period of time without fatigue. As an example of how little boost may be needed, I used 3 db of boost at both 50 cps and 12,000 cps and found that my editing efficiency improved almost 100%. Remember that this boost is intended for use only while editing; do not use it in rerecording or airing unless you do so purposely. Remember, also, that excessive bass can induce fatigue very quickly: balance the sound to your own comfort for the longest period of time and hearing fatigue will be reduced to the minimum.

There is no way I can describe to you how <u>you</u> would feel if <u>you</u> had hearing fatigue. That's something you have to find out ~~by~~ for yourself. I can tell you, however, what <u>my</u> reactions to hearing fatigue are: my ears feel "tight" and my perception of time and pitch are false, although I don't know it at the time - I find out later, after resting. Low-frequency sound appears "flat", with little resonance; for example, you could see the drummer hit the bass drum but would hear only a a sort of "blatt". High pitched sound appears shrill - almost ear-piercing. Some of the resonant sounds we call "wolf" tones will cause discomfort, almost pain. When I am extremely fatigued I know it's time to rest when I don't even <u>hear</u> a sound that's on the tape., if it is a short sound. Hearing fatigue causes temporary deterioration of hearing; you shoul avoid it by resting and by making your equipment give you the kind of sound you can listen to comfortably.

The ways in which hearing fatigue affects us is of more than ordinary interest to the tape editor because they exaggerate~~s~~ the normal limitations of hearing and point~~s~~ the way to several sophisticated techniques of tape editing that take advantage of these limitations. Some of the errors of perception present to a high degree during auditory fatigue are also present, to a lesser degree, during normal, unstrained, listening.

<u>Time factor in hearing.</u>

Most authorities on hearing agree that recognition of pitch at mid-frequencies can be made in 1/100th of a second, with the time needed increasing for lower and higher frequencies. In other words, it takes someone with normal hearing somewhat ~~XEEX~~ more than ~~1/000h~~ 1/100th second to recognize a musical sound - a sound having a definite pitch. If the time is too short for recognition to take place, he hears the sound without definite pitch - as wide band noise. If the time is too short for this to take place, he hears nothing. To translate these rules to actual tape editing, let us say you are cutting ¼" tape in an Editall Block. You always use the diagonal slot for cutting the tape and this cut~~...~~ is made at 45 degrees on the tape. At a tape speed of 15 ips the 45-degree angle cut makes the splice~~, at the cut,~~ 1/60th of a second long. That time is long enough for pitch recognition to take place at all audible frequencies. Of course, editing at 7.5 ips, the time would be twice as long, or 1/30th of a second, and so~~x~~ on at even lower speeds, which, by the way, make editing too difficult for normal work. Never cut sound tape at 90 degrees; you would hear noise at the cut if you were to do so, noise created in most cases by the too abrupt ending of sound. You can prove this to yourself, if you wish, by the following: Isolate a few seconds of tape recorded from any constant musical source - a pitchpiee, oscillator or even a whistle will do - between two lengths of leader tape, with the splices made perpendicularly at 90 degrees. Add another recorded tape, with the splice to leader tape made at 45-degrees. You will hear noise when playing back the 90-degree splices, no noise with the 45-degree splices.

Hearing persistence.

Hearing persistence is the sensation of hearing continuing after the sound that stimulated the sensation has died away. I have never yet been able to determine whether this effect is due to reverberation within the physical ear, time effects in the nerve system or lag in recognition in the brain. However, persistence has been measured and found to be factual. Scientists say that it varies appreciably with intensity of sound. In normal hearing, at sound intensities commonly encountered, the persistence effect lasts a very short time. I have found that it cannot be depended on for much longer than .03 to .04 second. This is perfectly natural and desirable, since music as we know it would not be enjoyable with longer persistence times.

Hearing persistence can be used in tape editing, not always but occasionally, to make fairly good transitions on edited tape from one segment of tape to another without rerecording. ~~IX~~ This blending may be done most easily following applause, which is in turn followed by speech or music. Cut the tape right after an isolated clap of applause and splice to a following word or note, leaving no space at all between. When this operation is performed properly it will sound as though the applause continued for a split second underneath the voice or other following sound.

Memory shock, or obliteration.

What I call "memory obliteration" is an editing trick similar to the above noted persistence blend. It ~~comes in handy~~ IS USEFUL when there is either no time to rerecord or no equipment for rerecording on hand and when there is a noticeable difference between the background sound of two tape sections that are to be joined together. The insertion between the two sections of any slightly shocking sound - a cough, a mike noise, a door slam or ~~xxtk~~ any characteristic noise - will, in most cases, cause the listener not to notice the background difference. He will accept the whole tape as perfectly normal. As I mentioned before regarding this kind of editing, do not cut in so-called "clear" tape; cut immediately after sound, or before sound, but be consistent in YOUR cutting METHOD.

In using any of these techniques (persistence or memory shock) do not consider the job well done unless, after a rest period, you can listen to the edited sequence without hearing the splice. It must sound natural-as if it happened that way. An imaginative tape editor, for example, in cutting from heavy applause to a following voice, might cut part way into the first following word, on the valid assumption that if the sequence had actually happened the way it was edited, part of the first following word would have been drowned out by the applause.

"Third Sounds"

If you have been editing tape while affected by hearing fatigue, you may notice, AFTER ~~while~~ LISTENING AND listening to the tape, that there may be an occasional sound that is difficult to analyze. This sound is the result of editing too closely, so that one sound followw another with not enough space betwwen to allow time for recognition to take place. This unintelligible sound, which I have named a "third sound" can always be cleared up by taking the splice apart and inserting a minimum of 1/60th second of background sound from the same sequence.

SECTION II

Reverberation in tape editing.

The commonest use of reverberation (artificially rerecorded) is for dramatic effect. For example, in most people's minds a ~~oration~~ speech delivered in a reverberant mode is presumed to have been ~~given~~ GIVEN in a large auditorium. Thus, when we want to place a voice in a large auditorium, we rerecord it after editing with appropriate reverberation. Many large auditoriums today are quite free of reverberation, especially when filled, but if we are to produce the right "picture" in the mind of the listener, we must cater to his association of reverberation with large auditoriums. Things, in this world, change much more rapidly than ~~ideas and~~ OUR preconceptions.

A second use of reverberation is to improve the intelligibility of recorded speech. If two identical tapes of the same intensity (or level) were to be reproduced through the same loudspeaker alternately, the reverberated tape would appear to be louder. Also, due to other effects, reverberated sound would seem more brilliant and easier to understand, provided that reverberation time is quite short. Consequently, a good way to improve a tape that HAS been over-filtered (to reduce noise, for example) is to reverberate it slightly in rerecording. In many cases this measure produces an intelligible tape out of what otherwise could not be understood. It is up to the recordist to determine the reverberation time, an~~s~~d amount, to be used. The criterion is: which method makes the speech ~~most~~ BE intelligible? In a well-equipped tape editing room, notching filters, roll-off filters and reverberation equipment should be available so that filtering and reverberation can be effected simultaneously with rerecording.

Another technique which utilizes reverberation is to match one sound with another, ~~both~~ each from different origins. For example, a speech recorded by a celebrity in Hollywood was played back over a public address system in New York as part of a memorial dinner. The next day this speech, including the New York audience, was to form part of a broadcast. Part of the preparation of this program consisted in reverberating the original recorded speech (from Hollywood) to approximatelly the same reverberation time as that of the rest of the recorded New York program, mixing the speech with audience recation and applause and rerecording the whole ~~thing~~ tape. In this way, instead of a voice obviously emanating from loudspeakers, the speaker sounded as if he were present at the New York dinner and did not sound incongruous in the least. All these techniques for altering sound make it possible for actors, if necessary, to record bits, wherever they happen to be, on tape or film. The editor can then match all the bits and edit them together into a finished production. Of course, this would necessitate duplication of scenery (FOR FILM) and might not be economical - but it is possible.

In order to edit sound that has been recorded in a reverberant atmosphere it is often necessary to use artificial reverberation. This comes about because, with longer than normal reverberation times, the reverberating sound often forms PART OF the background sound of the next word or note of music. For example, in speaking in an auditorium with a reverberation time of, say, two seconds, an inexperienced speaker would talk more rapidly than he should, with the result that each sound he made would start a train of reverberations lasting two seconds. An experienced speaker, realizing this, would cut his speech drastically and speak slowly enough so that his speech would not be disastrously garbled. However, in both cases, these speeches cannot be cleanly edited because a phrase cut out would still leave its echo behind it. ~~so to speak~~. The only way approaching good technique is to edit and then reverberate the edited speech, thus tending to override the original echoes with ~~the~~ new ones superimposed during rerecording with reverberation. Even this, however, will in most cases not produce good intelligibility but would be more believable.

In editing music we are sometimes confronted with a situation similar to the above. It may not even be reverberation that causes the difficulty but the natural resonance or "overhang" of the instruments themselves. The editor may then cut at a rest, in tempo, when reverberance and overhang have ceased, or within sound, preferably, on two identical sounds, in precise rhythm. If it should prove impossible to cut according to one of these two methods, either of two other techniques may be possible. One way is to cut after the last sound that one wants to include and just before the sound that is to be omitted. In other words, you are eliminating any overhang or echoes. Then splice blank tape to this sequence and rerecord it to fresh tape through a reverberation unit or chamber, fading in (turning up the level control through the reverb unit) just before the last note. The tape may be marked to help you fade in at the right point. Now this bit of rerecorded reverberated sound can be edited into its proper place in the sequence. If this technique has been performed correctly the cut will not be observed, not even by the musicians who played the original music.

The other technique is to make the needed cut, splicing in enough blank tape to prese the tempo, and then to rerecord the whole sequence through the reverberation faciliti increasing reverberation where needed. It is best, when using reverberation in this manner, to pay less attention than usual to the level-indicating meter and to trust more to your hearing.

~~[illegible]~~