ultrasonics. The audio signal is used to modulate an ultrasonic oscillator. The modulated ultrasonic signal is passed through a spring which reflects its signals back and forth until the energy is lost, and each time the signal hits the receiving transducer some of it is passed to a demodulator which removes the ultrasonic carrier and passes the resulting audio repetitions to the system output. Since the carrier is at 20 kc, the maximum band over which the spring has to work extends in our example from 20,000–9,000 or 11,000 to 20,000 + 9,000 or 29,000, the sideband frequencies for the highest (9,000-cps) audio signal. This is a ratio of less than 3 to 1.

Fig. 7-5 is a schematic diagram of the single-channel Panoramic Adapter. Audio input signal at the IN terminal, suitably attenuated, goes to the grid of triode V1, thence to the output. This is the direct signal, which is always present with reverberated signal in these systems.

V2 is the 20-kc oscillator, its inductor the primary of transformer L4. V3 is an ultrasonic amplifier feeding broadly tuned coil L1, from which (through a .01-μf capacitor) ultrasonic signal goes to the input crystal transducer X1 of the

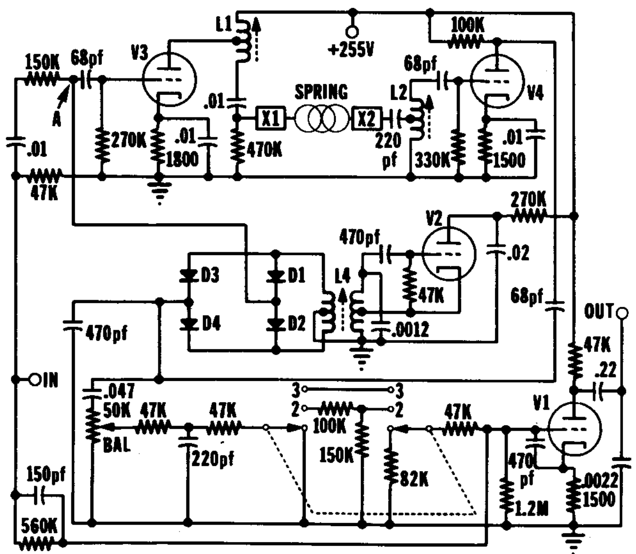


Fig. 7-5—Schematic of a single-channel Baldwin Panoramic Sound system using ultrasonics.

spring delay line. Since ground is at the center of the L4 secondary and both ends are symmetrically above ground, the load which receives the 20-kc generated signal can be assumed to lie between ground and the junction of D1-D2, which have finite and equal resistance even when conducting due to the 20-kc signal.

In the absence of any audio signal at the IN terminal, the two halves of the secondary of L4 and D1-D2 form a bridge circuit, the same potential always being at the junction of D1-D2 and ground, which is the coil centertap. No 20-kc carrier is therefore placed on the V3 grid, so that lacking an audio signal the carrier is suppressed. When audio is applied, D1 conducts on the negative alternations and D2 on the positive, the conduction being proportional to the instantaneous audio amplitudes. At point A, therefore, there appears both the audio and clipped negative and positive cycles of 20-kc signal corresponding in polarity and amplitude to the audio wave. These 20-kc alternations are clipped, of course, due to the diode action. The 20-kc signals pass through the 68-pf capacitor to the V3 grid and V3's plate passes them to L1, which removes the upper harmonics added by clipping and passes the signal to transducer X1 as a double-sideband suppressed-carrier signal.

X2 is the receiving transducer, connected to another broadly tuned circuit L1 and amplifier V4. From the V4 plate the signal goes through a blocking capacitor to diodes D3-D4, where the carrier is reinserted and the signal demodulated. The signal, which now consists of the audio plus the 20-kc carrier, goes to a balance control, which is a preset adjustment of maximum reverberated-signal level compared to direct signal. From here it passes to a 3-position, 2-circuit switch controlled by the organist. The switch in the position shown kills reverberation. In the other two positions it transmits differing amounts of reverberated signal to output tube V1. Between the junction of D3-D4 and the tube grid there are capacitors to ground which filter out the 20-kc carrier signal.

A 2-channel Panoramic unit is also available. It has a single oscillator and L4 which feeds two sets of the remaining circuits, including two springs.

The Schober Reverbatape Unit

A spring reverberator has the great advantage of simplicity and resulting reliability, but most people do not seem to feel that it gives a particularly realistic effect. Springs have their own resonances and sometimes even twang, especially when shocked by application of a sudden tone. At least the existing systems do not reverberate the lower and higher frequencies, and the existing methods of controlling apparent reverberation time are not parallels of the real thing. The worst offender is the small Hammond unit, at least for organs and quality music systems; it is often used today in guitar amplifiers, where its rather odd effect may be good since the popular guitar combos seem to look for out of the way sounds, including deliberate distortion. But any of the three units described is better than nothing if used in judicious moderation, so great is the enhancement afforded by any suggestion of reverberation. The larger Hammond unit and the Baldwin panoramic are quite beneficial to organ sound.

The Schober Reverbatape Unit is a development of my own, begun several years ago in an attempt to make a reverberator with a definitely realistic effect. Thousands of them are out in the world now, on both Schober and other organs, and I think I can unblushingly report that no one has ever seen fit to deny that its effects are more realistic and satisfying than anything else available to organ owners.

The Reverbatape Unit is a small, special-purpose tape recorder-reproducer. The use of tape provides the complete controllability of repetition rate, decay rates, and reverberation time *vs* frequency which is nearly impossible with springs. The idea is far from new, as tape reverberators have been used for years in professional recording and broadcasting, but the units have all been highly expensive and rather large. The problem was to make a device at a relatively low price but with the kind of reliability needed for use by laymen. If the following description tends to sound rather enthusiastic, remember that Daddy is talking about his baby. Quite frankly, over the five years or so the Unit has been on the market there have been some problems with mechanical reliability. In the latest model, however, cleaned up by Robert C. Avedon, a far better engineer than me, these problems have been thoroughly licked, as more than a year and a half of use in a great many homes has shown.

A Bird's Eye View

There is a continuous loop of tape which constantly passes through the machine, running in turn over a recording head, three playback heads, and an erase head. Program material to which reverberation is to be added is fed to the input of the Unit.

The output of the Unit, which is fed to the usual power amplifier and speaker system (or, depending on the application, to recording or broadcast circuits) contains first the original program material merely passed through the Unit without change of any kind. Second and more important, it contains program material which has been recorded on the tape, then picked up by each of the three playback heads in turn. Since the tape takes time to get from the record head to each playback head (about .11 second from each head to the next), each playback-head signal is a delayed repetition of the original material.

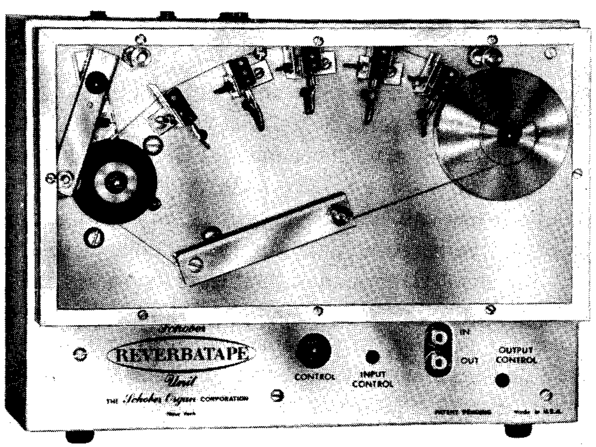


Fig. 7-6–Schober Reverbatape Unit is a small box of heavy steel, tape mechanism on face.

The third playback head, in addition to feeding the third delayed repetition to the output, also feeds a signal back to the record head, so that the process is repeated for another tape pass across the heads. The result is a series of equally time-spaced repetitions, each repetition decreasing in amplitude by a calibrated amount from the previous one until finally the signal is too small to be heard. The effect is that of an auditorium where repetitions are obtained by bounces from the walls, the sound losing some energy at each bounce, until nothing audible is left.

The user controls the effective length of the decay, and thus the apparent characteristics of the "auditorium," with a single-knob Reverbatape Control which varies reverberation time (time for a 60-db decay) from zero–no reverberation–to over 6 seconds.

The Reverbatape Unit consists principally of a tape-handling mechanism built on the front face of a heavy steel box, shown in Fig. 7-6. The tape itself is a very special kind. The oxide is dispersed in an unusual binder which has enormous holding power, so that continuous passing over magnetic heads causes very little deterioration. The small amount of oxide rubbed off comes away as a dry powder, which simply falls, rather than as "gunk" which would remain on the head gaps and foul them as with the usual tape.

The tape is in the form of a single continuous 19-inch loop, guaranteed for 500 hours of operation. In laboratory tests single loops have been run for over 1,000 hours without audible degradation. A standard splice, made with pressure-sensitive splicing tape, is unsatisfactory because the adhesive cold-flows and the splice slowly pulls apart. The tape is formed into loops by a welding process.

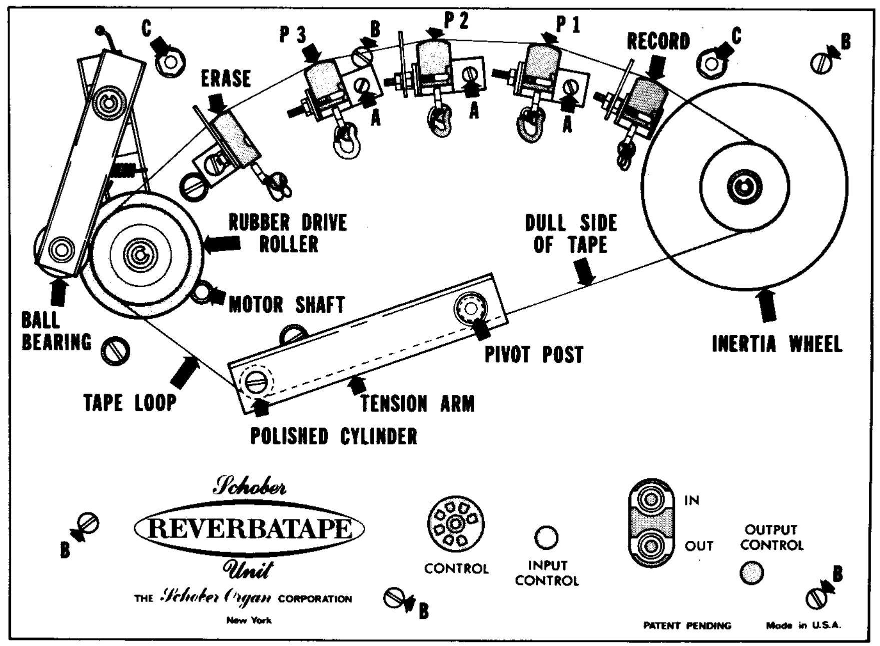


Fig. 7-7–Details of the Reverbatape mechanism. B: Screws holding printed board inside chassis. C: Screws for external head cover. P1, P2, P3: Playback heads. A: Head azimuth adjustments. Entire tension arm swings freely from pivot post to supply tape tension.

Tape Drive

The tape is driven by a small 4-pole motor within the box, whose steel shaft emerges as shown in Fig. 7-7. The rim of a rubber drive roller on an arm is very

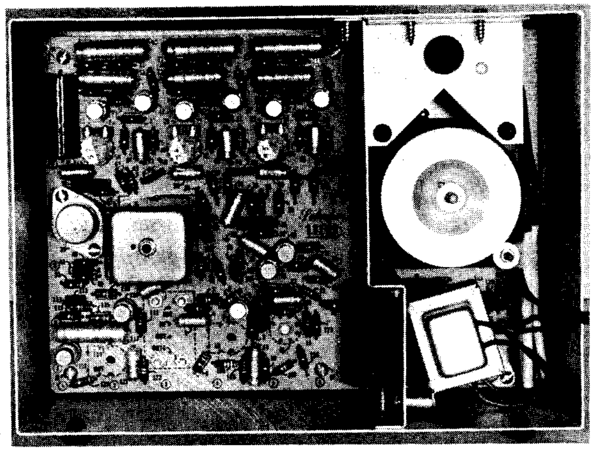


Fig. 7-8–An etched circuit board inside the Unit holds all electronic components.

lightly pressed against the motor shaft by a wire spring, and the tape rides over a smaller-diameter section of the roller. For positive drive, the tape is pressed against the roller by a ball bearing on another arm under spring pressure.

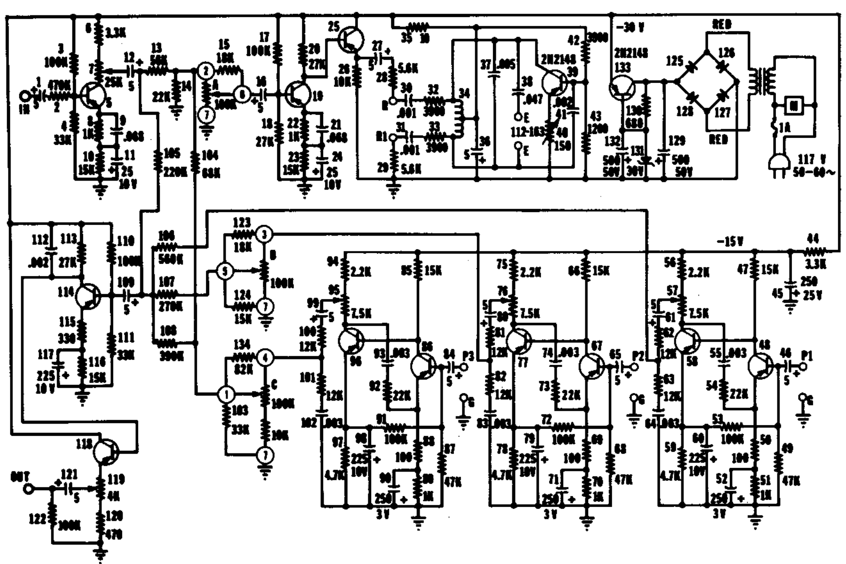


Fig. 7-9–Full schematic of the Schober Reverbatape Unit. The latest bias-erase oscillator circuit is slightly different from the one shown.

When the machine is not running, the pressure of the rubber roller rim against the motor shaft is light enough so that no permanent indentation is made in the rubber by the shaft. However, when the machine starts, the fact that the roller is pulling the tape against some resistance tends to snub the roller against the motor shaft. The roller, in effect, pulls *itself* up tight against the shaft. Also, as the motor shaft drives the rubber roller, a force is created which tends to swing the arm on which the roller is mounted *toward* the motor shaft, further pressing the roller and shaft in firm contact. This is a "self-energizing" system, unique in tape mechanisms to our knowledge, which does away with any need for mechanical disengagement of the two when the Reverbatape Unit is shut off, and considerably simplifies it.

A tension arm near the center of the mechanism is pivoted at one end and has a highly polished cylindrical surface at the free end. The weight of the arm pressing down on the tape provides just the right tension to hold the tape in firm contact with the magnetic heads without pressure pads. After the tape passes beneath the tension arm, it rides over the inertia wheel at upper right. This precisely machined brass wheel runs freely on its shaft and smooths out any variations in speed. It is a flutter filter.

The tape runs over the record head, three playback heads, and an erase head, in that order, at a speed of about 12 inches per second. The separation between heads is such that a given point on the tape takes about 110 milliseconds to go from one head to the next. After passing over the rubber drive roller, tension arm, and inertia wheel, the tape again passes over the record head.

Circuitry

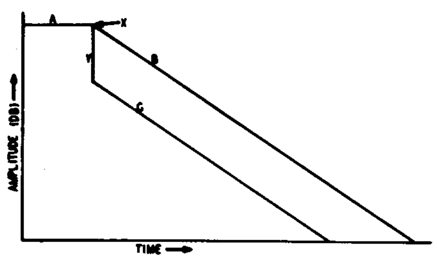
Being essentially a tape recorder, though a very special one, the Unit requires the usual stages—record amplifier, bias-erase oscillator, a playback amplifier for each head—plus special stages. All the circuitry is built on a single printed circuit board which may be seen in the rear-view photo of Fig. 7-8. Fig. 7-9 is a complete schematic diagram. An input level control (potentiometer 7) is provided on the board so that the Unit can be adjusted to place maximum undistorted recording level on the tape, whatever the available input voltage may be, to get maximum signal-to-noise ratio. The input control is in the collector circuit of input transistor 5, and the input system as shown can be adjusted for full recording when the input source maximum is as great as 3 volts or as little as 0.3. Actually, the stage has a 26-db gain, but input resistor 2 results in a 26-db loss, giving unity gain between input terminal and collector. Resistor 2 provides a high input impedance for the system, useful for many applications. However, if the available signal maximum is less than 0.3 volt, resistor 2 can be reduced so that maximum record level will still be reached. The sacrifice is, of course, input impedance, which comes down to a minimum of 22,000 ohms when resistor 2 is zero ohms. In that case, input maximum can be as little as .015 volt.

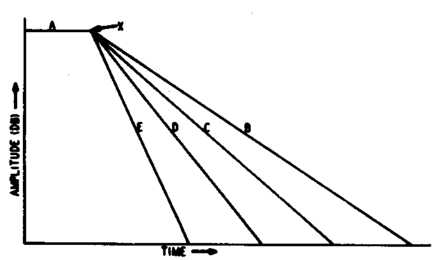
The Unit also has a level adjustment 119 at its output. In most applications this is set to that output is the same as input. However, since each control has a 20-db range, the Unit can usually be set to give either a loss or gain if wanted.

The input stage has most of the emitter resistance bypassed (11) sufficiently to keep the response flat to well below the audible range. The 1,000-ohm part (8) of the emitter resistance is bypassed by 9, which is small enough to result in a response at control 7 which begins to rise at 2 kc and reaches +12 db at 10 kc. This preemphasis (which is cancelled in a later stage by a complementary deemphasis) is instrumental in reducing the hiss at the output of the Unit to a very low level.

The signal leaving the input stage is applied simultaneously to the output stages 114-118 and to the recording system. The signal passed to the output stages eventually emerges at the output unaltered in any way. This is called the *direct signal* and is the original sound of the program source.

The signal going to the recording channel is first passed through a voltage divider (13 and 14) before going through control A (one section of the external 3-gang Reverbatape Control) and finally reaching the base of transistor 19, the recording preamplifier. The voltage divider adjusts the signal level to that required at the base of transistor 19 and provides the proper source impedance for control A.





Figs. 7-10 (left) and 7-11 (right)—Graphs show realistic and unrealistic sound decays.

The recording preamplifier stage 19 is virtually identical to the input stage. Capacitor 21 provides preemphasis as in the input stage, but for a different reason—to compensate for the high-frequency loss of the playback heads.

The output of transistor 19 is direct-coupled to the base of transistor 25, the recording amplifier. This stage operates as an emitter follower with the record head (terminals R and R1) connected from emitter to ground through suitable series resistance 28 and 29. The series resistance is sufficient to cause constant audio current at all frequencies up to about 10 kc, dropping off to −3 db at 20 kc.

Each playback head feeds signal to a playback amplifier, transistors 48-58, 67-77, and 86-96. The amplifiers are identical, each having d.c. negative feedback which stabilizes them thoroughly and raises the input impedance so that with the 90-mh heads used (Nortronics) there is no treble loss below 20 kc and the 5-μf input capacitors allow flat response down to 20 cps. Output from each amplifier is taken from a miniature potentiometer (57, 76, 95), provided to allow factory calibration compensating for head production tolerances. (Mechanisms of kit Units are calibrated in a special jig and the preset, marked pots go into the kit.) Resistor 63 and capacitor 64 and their duplicates in the other two channels provide a slow rolloff amounting to about 6 db at 10 kc, to simulate the reduction in reverberation time in auditoriums, where highs are always absorbed more rapidly by the reflecting surfaces than lows.

The output signal of the first playback amplifier is passed directly to the mixer, transistor 114, through 106. The outputs of the second and third playback amplifiers are passed through reverberation control sections B and C and resistors 107 and 108 to the mixer stage. The direct signal is also mixed into this stage through 105.

The mixer stage is basically the same as the input stage with the exception that high-frequency *deemphasis* takes place here to offset the preemphasis of the input stage. Deemphasis is caused by 112, which shunts the collector resistor at high frequencies.

The output stage of the Unit is emitter follower 118. This stage is directly coupled from the mixer, its stability being controlled by that of the well stabilized mixer. The output impedance of the Unit depends on the setting of the output control 119, varying from about 50 to 1,000 ohms.

Control System

The control methods usually used in artificial reverberation systems—especially the spring systems—are unrealistic. Usually there is a direct channel, through which sound goes without added effect, and a parallel reverberation channel, in which reverberative repetitions are added to the original. These are combined at their outputs. To reduce reverberation time, a control simply reduces the gain of the reverberation channel, so that relatively more direct sound is heard.

Fig. 7-10 shows the result. With full reverberation setting, the direct sound is represented by line A. The sound source ceases at time X. Then the reverberation channel causes a slow decay along line B.

When the reverberation channel is turned down, the direct sound A stops at time X. Then there is an abrupt drop in level (Y), after which the decay occurs at the same rate (C) as before. No real auditorium does this. Instead, the decay always begins smoothly just when the sound source is removed. Fig. 7-11 shows this effect at various total reverberation times. Each decay curve B, C, D, E, is a smooth one beginning exactly at point X, without any abrupt drop after cessation of the source. The decay slopes are, of course, different. To accomplish this necessary bit of realism in the Reverbatape Unit a new kind of attenuation system had to be devised.

Fig. 7-12—Reverbatape Control is a 3-gang potentiometer, can be mounted on wood or metal panels or in available plastic box. Decal is provided for markings.

The Unit is so calibrated that when the external control, Fig. 7-12, is at MAX, the output of the Unit due to the first playback head is about 1 db less than the output due to the direct signal. Output due to the second head is 1 db less than that; and signal due to the third head is still another decibel lower. Output of the third head is also fed back to the record amplifier through resistor 104, Fig. 7-9, and rerecorded, so that this cycle continues—each succeeding repetition 1 db less than the last.

With this control setting, and with the repetitions spaced .11 second apart, it takes 60 repetitions or 6.6 seconds for an attenuation of 60 db, and this is the maximum reverberation time of the device.

Faster music or music requiring greater clarity requires shorter reverberation times. These are obtained when the control is turned counterclockwise, by increasing the attenuation between played-back repetitions. For example, when the setting gives 2 db between repetitions, only 30 repetitions or 3.3 seconds is needed to reduce level by 60 db. The control is continuous from maximum down to zero. The following explanation of how it works requires careful attention to avoid confusion.

Maximum reverberation time is obtained with the sliders of controls A, B, C all the way up. (These controls are ganged on one shaft and available to the organist.) Control A then sends full signal to the recording amplifier 25. Signals from the three playback amplifiers join at the base of transistor 114. With the Control all the way up, the calibration potentiometers 57, 76, and 95 have been so set at the factory that the signal reaching 114 due to head 2 is 1 db less than that due to head 1; and the signal at 114 due to head 3 is 1 db less than that due to head 2. In other words, the outputs due to the three playback heads are 1 db apart.

Let us specify certain terminology. Let us always refer arbitrarily to the level of signal appearing at the Unit's output terminals *due to passage through the direct channel and not from the recording system* as 0 db. With the Control at maximum, the three calibration potentiometers have been so set that output signal due to head 1 is at −1 db, that due to head 2 is at −2, and that from head 3 is at −3 db.

Signal from playback head 3 not only goes through resistor 108 to join the other signals at the base of 114; it also goes through resistor 104 to control A for rerecording or feedback. This signal appears at the control along with the signal from the Unit's input. However, the signal appearing at the control from head 3 is 3 db lower in level than the signal due to the Unit input. This 3-db-less relationship, which is determined by choice of the components, is always present. Since the signal obtained at the base of 5, recorded and played back through head 1, resulted in a system output of −1 db, the *fed-back* signal produces a system output due to head 1 which is 3 db less than that, or −4. This is 1 db less than the output due to the original head 3 signal, and it continues the scheme of continual 1-db-apart repetitions.

We may also consider that, due to the setting of calibration potentiometer 76, the output of head 2 has a permanent 2-db attenuation, and that of head 3, due

to the setting of 95, has a permanent 3-db attenuation. Adding 3 db to each of these, for the second round of signals produced as the result of the feedback gives 3 db less than the original −2 for head 2, or −5, and 3 db less than the original −3 for head 3, or −6. Thus the action continues, repetitions at the output always 1 db apart.

So far the three ganged control potentiometers have always been at maximum. Before moving them, we must explain that they have special tapers, so that in any position the attenuation given by control C is always twice that (in decibels) of controls A and B, which are identical. Thus, if the Control is set so that A attenuates its signal 3 db, then control B has 3 db of attenuation and control C attenuates by 6 db.

Let us now reduce the Control setting so that A and B have 1 db of attenuation each—and C has 2 db.

The input signal passes through control A and suffers 1 db of attenuation. The output of head 1 is thus 1 db less than with the Control wide open. Instead of appearing at the Unit output at −1, it appears at −2.

The output of head 2 is also 1 db less than before due to control A. It also suffers another 1 db of attenuation in control B, making a total of 2 db. Therefore, instead of being at −2 as originally, it is now at −4.

The output of head 3 is also 1 db less than before due to control A. It also is attenuated 2 db more by control C, making a total of 3. So instead of appearing at the output at −3, it appears at −6.

You will note, therefore, that at this Control setting, the system outputs due to the three heads are, respectively, −2, −4, and −6 with respect to our standard 0-db direct channel. The reverberation time is reduced, but the decay curve, composed of equal decrements in repetitions, is still linear.

Now let us consider the fed-back signal with this Control setting. Since the signal to the recording head is attenuated 1 db by control A and control C attenuates the head-3 signal by 2 db more, the fed-back signal arrives at control A 3 db lower than it was before. In going to the recording amplifier it passes through control A, for 1 more decibel of extra attenuation. It therefore comes to the record amplifier attenuated by a total of 4 db more than it did when the Control was wide open. When the Control was wide open, the fed-back signal produced a Unit output of −4 due to head 1. Subtracting 4 db from that, the fed-back signal with the new Control setting, due to head 1, is −8 db, which continues the cycle of 2-db decrements as advertised. For a little mental exercise, you can postulate other Control settings and additional feedback cycles; you will find that decrements between repetitions are always equal.

Control rotation does not produce linear increments of reverberation time; there is a special over-all time *vs* rotation taper. The numbers on the Control dial do not indicate anything; they are just reference numbers. As the Control is advanced from 0, reverberation time increases rapidly up to a setting of about 1½ or 2 on the dial. As the Control is advanced beyond about 2, reverberation time increases much more slowly. The No. 2 setting on the dial gives about the minimum reverberation necessary for any effect at all. Thus, virtually unusable reverberation times are relegated to a small portion of the total rotation, while those which are the most useful are spread out over the Control range.

Reverbatape Installation

The Reverbatape Unit is essentially a voltage-amplifier stage, usually set for unity gain, with high-impedance input and low-impedance output. It can therefore be inserted in any audio system where these conditions are appropriate, and where the available signal-voltage maximum is something between 0.3 and 3 volts. (As indicated earlier, it can accommodate input signal maximums as low as .015 volt, but for this input the output voltage is still 0.3, and external attenuation may be needed.) A simple installation system, usually involving connectors and socket adapters has been worked out to fit each individual make and model of organ. The Unit is also useful for recording and home music systems. One version, called the RV-3C Reverbatape Tone Cabinet, sold through dealers, includes a power amplifier and speaker system with the Unit in the cabinet. It has six inputs so that as many channels can be reverberated; and the direct channel is removed from the Unit so that direct sound continues to come from the organ's individual channel speakers. A recent development is a 2-channel adapter for the basic Unit. This small printed board, optionally mounted inside the steel box, conveys direct signals from two channels to their individual power amplifiers without mixing them, and injects reverberations of both channels into both audio systems.

After any installation is connected, the Unit's input and output controls, available through holes in the front panel, must be adjusted. The input control is turned up so that the loudest possible signal just fails to overload the record head; and the output control is then adjusted for the desired volume. These adjustments are made by ear.

The Reverbatape Unit is available directly from Schober in kit form, with the mechanism assembled and the three pots precalibrated. It can also be sold by any organ dealer, completely assembled, with installation. The kit produces just the Unit itself, as it appears in Figs. 7-6 and 7-8 (plus the Control at the end of a 6-foot cable). The two dealer versions are the RV-3B—the Unit in a walnut box, the Control on the front, placed on top of the owner's organ—and the RV-3C, the Tone Cabinet setup.

Chapter 8 Tuning And Servicing

By and large, electronic organs are rather hardy animals, and the amount of repair service needed is satisfyingly small, considering the complexity of their design. This happy state has prevailed all along, and now that most modern organs lack those old hot vacuum tubes, with their inherent life limitation, potentially destructive heat, and high operating voltages trying to break down any breakdownable component, service trouble is even less prevalent. Regardless of whether one likes the purely musical performance of particular organs, almost every one of them has been designed with components of highly adequate quality and engineered, both mechanically and electrically, with skill. It would be a fine thing indeed if television receivers performed with anything approaching organ reliability.

Since perfection, however, like infinity, is an unattainable quantity, service technicians must and do exist. Ordinarily, organ service is handled by the dealer's service department; but more and more an independent breed of organ servicemen is arising. They are not numerous, but the work can pay well and can be interesting. Of interest to many of these men is the very fact that it is specialized, requiring knowledge on a subject not often written of in books or technical periodicals. Most radio-TV servicemen will not touch an organ—and usually they should not be allowed to do so!

Tuning

Most organs (spinning-wheel Hammonds are the notable exception) have self controlled electronic oscillators, either as control elements in frequency-divider chains or throughout the instrument. The self controlled oscillator which permanently remains tuned to its nominal frequency with enough accuracy to satisfy good musical standards has not been invented, though organ oscillator stability in general conforms to the highest standards of technical excellency. Inevitably, even though it may take six months to a year, every such organ needs a tuning. While a professional tuner (a piano tuner, for instance) can do the job if properly instructed on what screw to turn,* it is usually a technical man—the organ service technician—who is called on for the job.

The unfortunate part of this basic service job is that it is rarely done until the owner calls for it, and that is not nearly often enough. The human brain has a fantastic ability to tolerate a mistuned organ or piano which has gone out of

*I once had my own very expert piano tuner tune a frequency-divider organ. It took him about 10 minutes, and he expressed the wish that he could get jobs like that every day—at the usual piano tuning fee!