System Simplicity in Audio

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The trend toward duplication—or even triplication—of controls in a high fidelity system is looked upon by this author as unnecessary, useless, and inconvenient. Furthermore, he tells you what to do about it.

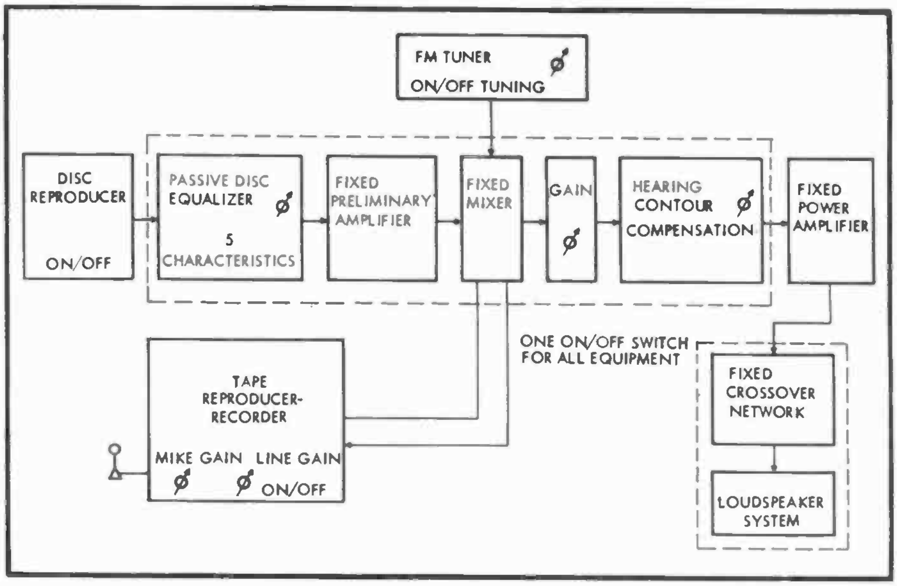

Fig. 2. Block diagram of ideal arrangement for an audio system.

THE MAJOR AIM of this article is to advocate "system simplicity" and to show that it can be employed to improve an audio system. Components recently featured in audio magazines show a trend towards increasing complexity. Although these gadgets are impressive looking, they require great skill to manipulate the myriads of knobs, My opinion is that many systems have overgrown and that the average audio enthusiast can achieve superior results by system simplification. Before system simplicity can be discussed intelligently, the object of an audio system should be known. My definition of a good music system is one that reproduces sound realistically—neither adding nor taking away, "Holding, as it were, a mirror up to nature." There must be no overemphasis of high or low frequencies. Concert-hall realism is not achieved by shaking windows with low frequencies or by hurting ears with high frequencies. I must add to the discussion of a good sound reproducing system, the plea that the listener attend as many live concerts as possible. Reproduction of sound may be pleasing but can never be more "real" than the actual performance. Moreover, the listener should have more opportunities to "keep his ear calibrated" by comparing the output of his system with "the real thing."

The current trend in the purchase of audio equipment is the selection of individual components by the audiofan who then assembles them into a system. *Figure* 1 is a block diagram of a system produced in this manner. This procedure has many disadvantages. First, there is no assurance that the components are electrically compatible. Second, there are too many controls which nullify or duplicate each other, Third, there are just too many controls. My conclusion after four years of buying or building components is that he who wishes to assemble a good system should obtain the guidance of a professional electro-acoustic engineer, one who can criticize objectively, who can test a system thoroughly, and who is not interested in selling any particular brand of equipment. The latter requirement is the most important. Then, a new system can be engineered to meet any needed specification, or an existing system can be simplified and improved. In either case the consultant will be able to prevent many errors and to provide system engineering.

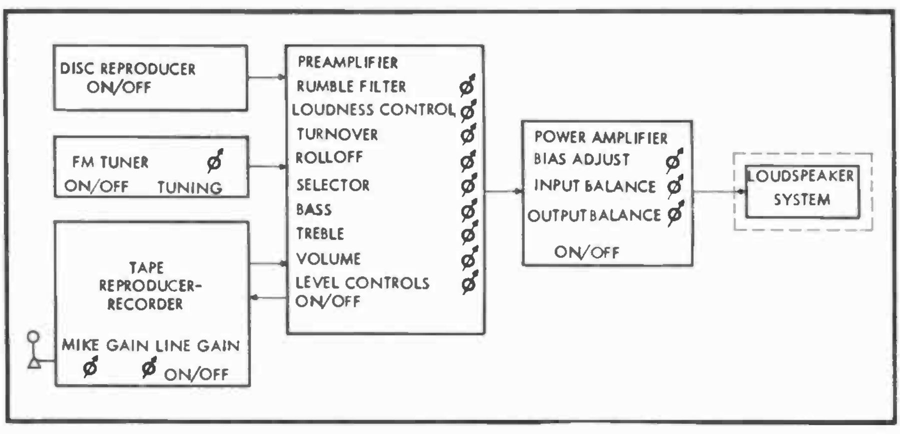

Fig. 1. Block diagram of system before simplification.

The ideal simplified system should be defined before I tell how my own system was simplified and improved. The general objective is to reproduce sound realistically at any level. The frequency response will cover the audible range, the power output will be adequate for the needs of the listening room, and the distortion will be inaudible. Controls and adjustments will be kept to a minimum, and where equalization is needed, fixed passive networks will be employed. From these generalities a block diagram (*Fig.* 2) of the ideal system can be made.

With the advice of the consultant, I devised the audio system described in the following paragraphs. Component circuit diagrams are omitted since conventional circuits are used. The number of controls was reduced to a minimum. Broadcast control room techniques were employed, and each unit was equipped with its own power supply and fuse. Individual parts of a unit were oriented for minimum hum. Standard telephonic techniques were employed in designing the wiring of the units. Source output voltages were set at fixed valves to meet input requirements of the Control Unit. The stepped Hearing Contour Compensator reduces the level by an amount shown on the compensator setting. The single gain control is therefore usable over its entire rotation, not just the first ten or twenty degrees.

An audio system is divided into three parts: sources, sinks, and controls. In turn, each of the three parts has subdivisions whose specifications and method of simplification now can be discussed individually.

Sources

Disc Reproducer. Disc reproduction is designated exclusively for 33⅓ rpm LP discs in manual operation. Since all of the program material which interests me is available on LP discs, only a single turntable speed is needed. I frequently wonder why turntable manufacturers do not produce a good single speed, 33⅓-rpm turntable. Elimination of the unwanted speeds, pulleys, and idlers is a form of simplification. A good quality magnetic cartridge completes the disc reproducer.

FM-tuner. The tuner has a.f.c. and a tuning indicator. Only two controls are necessary—the on-off switch and the tuning control. The output voltage was set at an average peak level of 0.5 v. rms. Equalized high-quality headphones can be plugged in for night listening.

Tape Reproducer-Recorder. This unit has NARTB equalization. Two gain controls are used in the "record" mode of operation: One in the "microphone" channel, one in the "line" channel. A VU meter is used. There is no gain control in "playback." Provision for both tape and input monitoring by headphone or loudspeaker is made. An Ampex 600 meets these requirements.

Sinks

Power Amplifier. The frequency response of the basic amplifier is flat within ±1 db from 20 to 20,000 cps. When the adult human ear is shown to hear beyond 20,000 cps, then I'll start worrying about extending the frequency response. Fixed compensation for loudspeaker characteristics have been added. The amplifier is designed to furnish one-half the 25-watt maximum power at 0.6 v. rms input, allowing three decibels of reserve power for possible overswing. The internal generator impedance has been adjusted by test for optimum damping of the associated loudspeaker system. A pulse from an RC circuit was used in this test, and the amount of inverse feedback was varied until best damping was secured.

Loudspeaker System. A warble oscillator and a calibrated microphone were used to set the proper balance between high- and low-frequency loudspeakers and to set the equalization for loudspeaker and room acoustics. This equalization is permanent since room acoustics change very little from day to day.

Controls

A commercial preamplifier was reworked into a "Control Unit." Two of the sources listed above were mixed by means of a resistive network into the preliminary amplifier at the proper levels according to the output voltage of each. Because each source has its own power supply, its output signal can be removed by simply cutting off the power. Thus the selector switch could be eliminated. However, a switch was provided for tape playback to eliminate a feedback loop during recording. Tape recording may be monitored by headphones or by loudspeaker.

Preliminary amplifiers with separate high- and low-frequency controls for disc equalization have a common fault. Although the low-frequency control is intended to effect only the low frequencies it also effects the highs, and vice versa in the case of the high-frequency control. Therefore, unless accurate calibration has been made of all possible combinations of those controls, front panel markings are far from accurate. Disc equalization has been simplified to a one-knob control. Five equalization curves based on published curves of representative disc manufacturers are sufficient for all discs (old records being taboo). Circuits were designed to switch resistors rather than capacitors to minimize the effect of switching transients. The RC values were calculated and then corrected by frequency measurements to yield the proper equalization. The "Rumble Filter" was removed entirely since a transcription turntable is used and unreasonable "bass boost" is avoided.

The main feature of the control unit is the replacement of conventional "bass" and "treble" controls by a "Hearing-Contour Compensator." The Hearing-Contour Compensator improves the realism of high-quality sound reproduction by compensating for the difference in level between the music produced in the concert hall and that reproduced at a necessarily lower level in the living room. It compensates for the variations in human hearing sensitivity to sounds of different loudness. The variations in hearing have been measured and have been found quite uniform for persons of normal hearing. They are shown in the Fletcher-Munson Curves of equal loudness. The principle of the Hearing-Contour Compensator operation is based on a study of the differences between Fletcher-Munson Curves, rather than on contour at any one acoustic level. Averages have been selected based on a series of subjective tests which included listening alternately to original sounds and then to the same sounds recorded and reproduced. Most of this testing was made with the orchestra of the Metropolitan Opera in New York and the U. S. Navy Band in Washington, D. C. The Hearing Contour Compensator performs in fixed calibrated steps of 0, 10, 20, 30 and 40 db. These figures indicate the difference in db between original and reproduced program levels. Appropriate attenuation is designed into the compensator. In case speech is reproduced, music equalization is completely wrong, since speech should be reproduced at about the same level as it was originally produced. For speech reproduction a switch is provided which retains the attenuation but removes the compensation.

It is important to digress at this point in order to speak again of the value of frequent listening to good orchestras under good concert hall conditions. Many "hifi" fans will find this an illuminating experience; a few may be discouraged and some will staunchly maintain that the orchestral sounds are much inferior to those reproduced by their "hifi" systems. This will prove the revised adage "be they ever so homely there are no ears like your own."

The number of controls in the system was reduced from twenty-one to ten.

CONTROLS

Before Simplification	After Simplification
FM tuning control	FM tuning control
Rumble filter	
Turnover	Disc equalizer
Rolloff	
Selector	
Volume	Gain
Loudness	Hearing Contour Compensation
2 level controls	
Bass	
Bias adjust	
Input balance	
Output balance	
Line gain	Line gain
Microphone gain	Microphone gain
5 on/off switches	4 on/off switches

This simplification of my audio system has had several broad results. I now have a positive knowledge of the average acoustic output of my speaker, whether at full volume or low, and the balance is correct for every level. There is no dependence on acoustic memory and no extreme overemphasis. Reproduced music sounds close to that I hear in the concert hall. The use of the Hearing Contour Compensator permits the use of the following operating technique. After a source is selected, a suitable listening-room loudness level is chosen. The Compensator is adjusted to the necessary setting (−10, −20 phms, etc.) to furnish this loudness level. The gain control is then rotated clockwise fully, fading in the desired sound smoothly with no acoustic shock to the listeners. •