# Introduction to the Low Noise, High Bandwidth, Zero Rumble, Vacuum Tube Modification Manual

Ike Eisenson, Audio Dimensions, Inc. — 2nd Edition 1977

## THE BOOK

This text was written by a multitude. For the past four years, audiophiles around the world have contributed to its pages; their ideas and practices have been correlated, refined, tested and retested, then combined with others and validated by the finest test equipment available, including the most dependable of all: the human ear.

The author has for some years acted as a clearing house for information on the use of vacuum tube equipment for the reproduction of sound. During that period, he has built more than one hundred amplifiers and preamplifiers. His goal was the construction of equipment which met a demanding set of standards. As a result, there appears in this book a great deal of supposedly technical information. It is expected that many readers will be far more technically qualified than the author, who is in many ways unqualified to attempt such a text. In the academic world of today, however, little attention is paid to vacuum tubes, and modern technology assumes that tubes are anachronisms. Those who find fault with the arguments within this book may have an education in solid state electronics which far exceeds that of the author, but it is likely that they were given only a most cursory look at tube theory, probably as an introduction to the world of solid state.

Virtually every advertisement for high fidelity equipment claims "realism" as one of the virtues of the hardware offered; the claims for a $15 phonograph cartridge are almost identical to those for a cartridge system costing hundreds of dollars, and the differences between a good speaker and a great one are subtle, as are the differences between any two pieces of expensive equipment. To the ear, however, only one criterion really matters: does it sound "real"? Therefore, the goal of the projects proposed by this text is the recreation of music, with the result being as close to what the microphones heard as possible. If one pays less attention to the "numbers" (distortion, bandwidth and power ratings) and more to the manner in which the sound is perceived, it is possible to fool the ear/brain system—and that should be the object of any fine music reproduction system.

Although this book may suffer from technical flaws, it is the sound we are after, isn't it? TO THE EAR, the principles are valid and the techniques work.

## THE ART

Since the first commercially successful sound reproduction system, man has steadily worked to reproduce more closely the original performance. At first, the economics of sound reproduction seemed to be the driving factor; it was less expensive to listen to a reproduction than to attend a concert. In the 1970s, however, economics underwent a significant change. Though tickets and transportation increased in price, it became unlikely that the cost of a first-rate audio system could be amortized, or written off, against money saved. As the price of equipment skyrocketed, it became obvious that factors other than entertainment were justifying the acquisition of fantastically complex and expensive music reproduction systems.

"Acquisition" is an appropriate term, as any psychologist can explain. Many audiophiles are obviously more involved in the quest for equipment than in a search for artistic achievement. To some, audio has been reduced to a set of specifications for which improvement is constantly sought. Others finger their knobs and switches, measuring a "system" by its complexity and pointing proudly at scope presentations and built-in power meters. Still others seem to relish the construction of components, and build kits or "scratch" units which, once complete, are seldom turned on.

There is a rare minority for which the equipment is a necessary evil. This group is less concerned with specifications than with the performance. A member of this group can be readily identified. When you enter his home he says "I've got a new DGG Rachmaninoff" rather than "finally got a new crossover". Further identification is provided by an examination of the system he uses; it will consist of a mixture of old and new, and bear witness to the owner's buying principles: "If it breaks, fix it. If it can't be fixed, replace it." This sort of audiophile probably does not subscribe to hobbyist magazines unless it is to read the record reviews. The constant claim to "state-of-the-art" does not even rate a glance, since his equipment works.

Regardless of the criteria by which one judges a system, certain truths remain inviolate. The purpose of the system, however simple or complex, is accurate reproduction of sound. When the system is working, listeners seem to fall into one of two categories: the first group listens for the resin on the bow, while the second is in rapture over the conductor's interpretation. The first group tends to assemble systems which are truly a sonic spectacle, capable of an 1812 0verture which breaks windows and leases.

One means of identifying members of this category of listeners is to count the number of times a technical adjustment is made during the play of a record. (A remote control is a dead giveaway, of course, even if it consists of a volume control.) The other category of listeners is interested in the gestalt, the realism of the "whole thing". It is seldom that a member of the audience at a symphony hears "the resin on the bows," since the most favored seats hear what the composer and conductor intend: the gestalt. The tympani, for instance, is usually heard from a distance of at least one hundred feet. Average sound pressure levels, at a live symphony, are less than 80 dB at a typical listening position, though peaks go much higher. Consequently, the real purist is likely to be unhappy if an audio system makes the tympani sound as though they were located within the loudspeaker, and may find 100 dB sound almost painful.

One of the paradoxical truths about audio is that these various and apparently contradictory positions can be resolved. It is possible to build a system which will please, if not delight, everyone.

The key to attainment of such a goal seems to be accuracy. Even the audio pervert, who likes bass drum strokes to move furniture, is likely to be a bit awed when he closes his eyes and cannot differentiate between reproduced and live music.

It is necessary, then, to discover those clues which the ear-brain system uses in distinguishing between a reproduction and "the real thing". A few simple experiments rapidly indicate that the advertised claims seldom deal with the really important factors. As an example, frequency response is definitely not one of the clues the ear-brain system uses to sense whether music is live. Proof is easy. Consider the man with a serious hearing loss, a man who cannot hear above 8000 Hz. Despite his problem, it is difficult to "fool" his sensory system. How about distortion? Two trumpets, by different manufacturers, and played by different performers, have a different sound, but both will sound "live". Real time analysis, with good equipment, will prove that the major difference is in the distribution of harmonic content. In fact, it is likely that either of the two horns will produce a waveform which appears to be a distortion of the product of the other. Yet both sound live' Reproduction of a solo performance, accurate except for the presence of limited harmonic distortion, will probably not be distinguishable from a live performance. Intermodulation distortion is not a major factor in the reproduction of a single instrument. If IM figures are reasonably low, that type of distortion alone should not permit detection of the electronic process of reproduction of even large masses of diverse instruments. In fact, there is a likelihood of IM in live performances, in the form of beat frequencies developed between co-located instruments.

Another highly touted characteristic of the "best" audio equipment is the residual noise level. It is a psychoacoustic fact that even quite high levels of hum and hiss are masked by the signal, and usually become evident only when one specifically listens for them. As for other types of noise—random pops and clicks, and a generally high background noise level—it is unlikely that the poorest system will come close to the noise prevailing at a concert or discotheque.

If all these factors are eliminated as contenders for the chief "clue" used by the sensory system in detecting whether sound is live or reproduced, what is left? There is a means of identifying the elusive remainder. Assume a system which reproduces music over an extremely wide bandwidth, say 30 Hz to 30 kHz, with low noise, and total distortion less than 1%. Listening to this unlikely system may or may not "fool" you, but it will certainly be a pleasant experience. At times, the system will probably sound extremely "live"; at other times it will not. If you could introduce a change which reduced the likelihood of fooling the senses, but did not change the bandwidth or distortion, it is probable that you would find one critical characteristic. A dynamic compressor/ expander, or "compander", might provide valuable insight as to just how the ear-brain system works. Used to expand the dynamic range of the program material, it is improbable that the music would seem "live" more often. Without a doubt, however, if it is used to compress the dynamic range, even slightly, you will know it immediately. This is one of the arguments used to support the candidacy of dynamic range as the sensory system's most important clue in identifying "live" music.

The very best of the companders is capable of expanding only relatively gross transients. That is, transients lasting more than a few milliseconds or so would be expanded, while those of shorter duration would be missed. It is the presence or absence of these very short-lived transients which is of such critical importance. If they are lost in the various electronic processes, no expander can recreate them. If a few remain, however, a compressor will remove them from the program material, as will overload at any point in the audio chain. It is contended, then, that a major factor in the reproduction of music is the recreation of "fleeting transients" of incredible amplitude (compared to the rms value of the rest of the program). When these transients are cut off, or "clipped", the process might generate high order harmonics, which are another clue to electronic reproduction. Frequency response, distortion and noise levels are also important to the enjoyment of reproduced music. If, however, one wishes to fool the ear-brain sensory/processing system, it is critical that these short-lived transients be present.

One of the advantages of vacuum tube equipment is its inherently high overload point. Many a fine transistorized preamplifier, for instance, can be overloaded by a high output moving coil cartridge playing through a solid state subpreamp. The result can be elimination of many transients necessary to accurate reproduction. On the other hand, a properly designed thermionic (vacuum tube) preamp is virtually incapable of overload by any available cartridge. When a triode vacuum tube does "clip," it generates predominantly second order harmonics, which are not as unpleasant as high order harmonics.

A simple test will indicate the transient handling status of a system. Select any of the test records which provide a band of pink or white noise. With a pin or razor blade, lightly scratch the record across that band. When played, the scratch should be higher in volume than the background hiss. With an oscilloscope, examine the output of the cartridge (conventional magnetic cartridges require a scope with high sensitivity). The trace should, between "pops", look relatively flat and of constant amplitude. When the stylus hits the scratch, the scope will display a pronounced peak. Measure the amplitude of the background noise, and that of the peak, and convert the two figures into a ratio. Once established, that ratio becomes a reference used in examining the remainder of the system. Next, using the same technique, look at the output of the preamplifier. A comparison of the output from a quite expensive solid state preamplifier with that from a common vacuum tube preamplifier indicates substantially more compression when the signal goes through the transistor chain. If the output of the preamp shows a transient/background ratio roughly equivalent to the reference figure, move on; otherwise, recompute the ratio. Examine the output of the electronic crossover, if used, but remember to add the HF and LF peaks. If no crossover is used, move on to the output of the amplifier. Finally, check the output of any passive crossover, again remembering to sum the signals of the LF and HF sections.

Another test which will provide a valid indication of the transient handling capability of equipment requires a square wave generator and an oscilloscope. The more vertical the leading edge of a square wave (at about 5000 Hz), the more likely that "fleeting transients" will pass through the device being tested. The conventional procedure for rise time measurement requires careful calibration of the scope, and a time base. For the purposes of this text, relative changes—measured on an uncalibrated scope—are deemed sufficient... and the use of tubes, especially triodes, will make musical infidelities tolerable.

## THE MARKET

A visit to any large audio emporium is sufficient to boggle the mind of the potential buyer of "components", particularly if he is making that critical transition from one of the commercial packaged units. The most influential factor in that decision seems to be friends, and the systems they have presented to our fledgling audiophile. A desire for "something better", and the purchase of one or two audio magazines, have led this customer into the clutches of one of the most avaricious groups in our country... the audio sales force' The neophyte, armed with his latest buyers' guide, is fair game. He feels he knows a lot about a relatively unsophisticated subject, and probably already knows what sort of equipment he wants to buy. He is concerned with amplifier power, distortion figures, number of speakers and their sizes, rumble, wow, flutter, noise and price. It would be difficult to convince him that a single eight-inch woofer and a small dome tweeter, in a well designed system, could sound more accurate than the MEGAHERTZ SPECIAL, with two big woofers and a five-way crossover.

The MEGAHERTZ SPECIAL may seem a bargain when compared to the smaller unit, and will certainly sound more impressive when first heard in the store. It will have more bass and more "presence", will handle more power, and will be bigger. When the salesman explains that "these speakers are bought direct from the manufacturer... we have an exclusive on this line", he may be saying that the speakers are a house brand. If he argues that the larger speaker.is cheaper because of a special marketing arrangement "eliminating the middlemen", that is another tipoff to the wary.

The best example of this sort of marketing is Apollo vs the Visonik. No experienced audiophile would prefer the former product, which has a list price much higher than that of the smaller but more accurate Visonik. In a competition between most house brand loudspeakers and the small Gale units, only the rock freak would fail to choose the relatively expensive Gale. The Advent loudspeakers compete directly in size and price with a multitude of house brands and obscure "special arrangement" units, yet the difference in profit to the dealer leads many to recommend almost anything but the Advent, which is one of the best bargains on the market today.

The same principles apply to electronics, though to a lesser degree, simply because of the relatively strict advertising standards imposed by various consumer protection agencies. To avoid a direct confrontation with "audiophile" components, some retailers (Olson, Radio Shack and others) offer only house brands plus a few sad examples of "fringe" products. Even a poor amplifier can be made to sound good when compared to a real piece of junk.

Cartridges offer yet another potential for rip-off. Only a small percentage of consumer cartridges are sold for "list price". A highly inflated list price has been imposed on the most popular consumer cartridge produced by Shure: the M9lED, at about $55. Since this cartridge has been available for years at between $15 and $18 to those "in the know", it is obvious that the $55 list price was created to permit a larger "discount" of package systems incorporating that cartridge. If the same approximate ratio of list and discount prices were to apply to the $75 V15III, an "audiophile" rather than a "consumer" product, that relatively good cartridge would sell for less than $25. A similar situation exists with other cartridges, including ADC, Pickering, Audio Technica, Grado, and more.

*Caveat emptor* never applied more than in the audio field; only automobiles and desert lots are sold with more hustle than a house brand package system in your local "discount" stereo emporium.

The experienced audiophile has probably already learned (painfully) about most of the more common deceptions in this field, and—by virtue of his reading this book—is likely to remember the advent of transistors in the hi-fi store. No technological "advance" to audio equipment was ever accompanied by more fanfare. Many Dyna, Marantz and McIntosh owners were immediately persuaded to trade for solid state, and never understood why they found themselves listening to music for shorter and shorter periods of time. The incisive, detailed quality of transistor sound was very attractive in the store, even when compared directly with the somewhat more liquid, softer sound of tube technology in the early sixties.

Today's tube technology is able to provide products with most of the sonic advantages of transistors, except for some of the very basic differences (life expectancy, heat, bulk and power consumption). The sonic comparison is remarkable, since most serious listeners rate tube equipment as substantially more accurate than transistor counterparts. Since tube devices generally cost more to manufacture than solid state items, there is a marketing problem: most vacuum tube components cost substantially more than comparable transistorized equipment.

When the audiophile is faced with a vacuum tube amplifier and a solid state amplifier at the same price, he will discover a tremendous difference in specifications. The tubed unit will be larger, heavier, and will have far less output power, more distortion, consume more power, and will present a continuous problem of tube replacement. The decision to buy the tubed unit will often result only from exhaustive listening, which most stores are unable to permit. Though it has been said that "ownership of any piece of audio equipment is typically a transient state", it is unfortunate that we must learn our lessons so painfully.

A Marantz 8b amplifier, of only 35 watts per channel output power, sold in 1963 for only $264. A good used 8b is worth more than that today. To build an 8b on a commercial basis today would require an ultimate retail price of more than $700... that figure buys a lot of transistorized watts. Since the majority of equipment is bought during a lengthy transition process from a Sears "stereo compact" to true audiophile equipment, that process can be quite expensive unless something happens to reduce the number of steps taken from one extreme to the other. It is worth arguing that the actual cost of a good Paragon system should include the dollars wasted in learning enough to buy one. Steadfast owners of old Marantz, Citation and Scott equipment must chuckle if they bother to read the advertisements prevailing today. The most overworked term in audio, "state-of-the-art", simply has no meaning to them. They take pity upon the man carrying a box marked "Supertuner MKII" from a store, who sees cases of MKIIIs being unloaded from a truck.

There is a lot of evidence to indicate that it is possible to stay with a given assembly of electronic components for many years, making modifications as technology, time and expertise permit. One important piece of evidence is a pair of Marantz 9 amplifiers which were recently modified in San Diego. The result has been favorably compared to (literally) the most expensive amplifiers on the market today, both solid state and vacuum tube. Vacuum tube technology can be applied to virtually any signal processing problem, including peripheral functions such as dynamic expansion/compression, noise reduction and equalization. For the most part, when these functions are required, they can be better performed by solid state devices. Accordingly, this text will concentrate on tube amplifiers, preamplifiers and power supplies of other devices. The basic contention is that the functions of voltage amplification, power amplification and frequency division can be performed (absolutely) better with vacuum tubes than by transistors, and (relatively) more cost-effectively with modifications than through purchase of new equipment.

# Chapter One

## Perfection ?

To establish realistic goals for any equipment modification proposal, it is first necessary to define "perfection". Obviously, the goal of any modification program must lie somewhere on the line between current status and that sublime, unattainable state to which every manufacturer lays claim in his latest advertisement. The term "state-of-the-art" (SOTA) attempts to describe the distance along that line that technology (and application) has moved. Unfortunately, SOTA for a particular parameter, or form of distortion, may be found in an otherwise unacceptable piece of equipment. Claiming SOTA status for a parameter such as harmonic distortion, then, is not valid. It is far more realistic to establish SOTA for a general function rather than for a single desired characteristic.

The search for "sonic perfection" has, in the minds of a series of inventors, been successful many times over the past one hundred years. An early mechanical phonograph used a friction device to transmit energy from a rotating drum to a diaphragm, and controlled the friction by means of a stylus/string arrangement. Claims for this mechanical amplifier rivalled those for another device intended to produce the same end result. The latter product used a supply of air pressure attached to a horn through a metering valve, and used the stylus to modulate the airflow, thus producing sound. Except for the ever-present hiss, this was certainly near perfection for its day. The term "perfection", then, is apparently subject to interpretation and is not nearly as absolute as one would like to think. For the purposes of this text, however, "perfection" will be clearly defined.

The perfect phonograph cartridge should convert the modulations of the disc groove into an exact electric analog, with zero wear of the disc and zero spurious signal content. This seems valid, but has little to do, apparently, with the way a real cartridge sounds. This conclusion can be reached by comparing the sonic characteristics of two cartridges with identical technical characteristics; in some cases, two cartridges, by the same manufacturer, with sequential serial numbers, will sound quite different.

The technical data published by the many cartridge manufacturers asserts that each of them has at least one product which is SOTA. The only thing these cartridges have in common is that claim, it appears, since there are many principles involved, including moving magnet, moving coil, moving shunt (variable reluctance), electret, strain gauge, piezoelectric, and variable capacitance. It is possible to find an example of each principle which has nearly the same technical statistics as the others, yet the resulting sound will vary significantly.

One means of determining "state-of-the-art" is to study the signal before it is impressed on the record (using the master tape, perhaps) and compare it to the signal produced by the cartridge. Though this procedure ignores the aberrations of a long string of electronics, a cutting head, and the disc itself, one would expect that the cartridge which produces a signal electronically closest to that on the master tape is the most nearly perfect transducer. This is, unfortunately, not the case. The perceived effect of each type of deviation must be weighed, of course.

The problem is determining the scale of values for each variation from perfection. The obvious answer—the sound—requires a subjective judgment, which is invalid as long as there are two audiophiles present. As a result of these arguments, which apply equally to virtually all components, we must conclude that it is not the oscilloscope which must be satisfied, but the specific listener whose opinion determines the relative merit of a device. If we assume that all really good cartridges are nearly perfect and try to identify those aberrations which are least tolerable, it becomes possible to separate the wheat from the chaff. These cartridges will all have good frequency response (bandwidth), and low distortion of all types. None will destroy a record rapidly, and all can be wired up to produce little or no discernible hum. One parameter which remains, as argued previously, is dynamic range. If we are searching for perfection regarding this characteristic,the chaff becomes quite evident, since all generators or magnetic cartridges are apparently eliminated from contention. This is due to two factors: mass and hysteresis effect. In any system involving a magnet, shunt, and coil, which move relative to one another, there must be movement before energy is generated. Further, once the movement begins there is a restriction to the rate of change of energy output, a characteristic of any inductance (coil) in the circuit. Once relative movement ceases, the energy remaining in the coil magnet system will dissipate through the circuit, thus preventing perfect decay time. These generators sense velocity, and below a measurable rate (frequency) will not produce significant output. The result is the development of the various equalization curves (now standardized: RIAA) that compensate for the difference in output at low frequencies (low rate of relative movement) and high frequencies.

From a philosophical viewpoint, one might say that the requirement for equalization (of the signal from the cartridge) utterly eliminates any magnetic cartridge from contention. Practically, though, if a "perfect" preamplifier were available, this would be a minor issue.

If the transduction options were explained to a competent physicist, he would be quick to identify the strain gauge concept as the most likely to approach perfection. The old Euphonics, the Panasonic, and the spectacular Win Labs cartridge all operate on the principle of a resistance which varies with strain produced by a moving stylus. Since the system senses amplitude rather than velocity, it is capable of response down to DC, or a permanent deflection of the stylus. In sophisticated electronic circles, a bandpass of the mere audio range (20-20kHz or so) is no challenge at all, due to our gigahertz technology.

The limitations of the strain gauge system should not be the electronics involved. In fact, solution of the electronics problem is a simple affair, as will be described later in this text. The real limiting factor in a strain gauge cartridge appears to be the stylus assembly and the coupling method. The most successful techniques in these areas appear to be applied to the Win Labs product, which, in its early stages, was plagued with electronic (solid state) problems. Though most experienced audiophiles are accustomed to variation among apparently identical cartridges, it now appears that both the Panasonic and Win Labs are outstanding, with the latter product recently taking a decided lead in both performance and consistency. You may have noticed that the word "cartridge" was used, rather than "system"; the solid state electronics of the Win is far superior to any previous product (February 1977), but the Panasonic electronics are conically mediocre. The author contends, then, that the power supply and preamplification problem$ posed by the Win Labs and Panasonic cartridges are best solved with vacuum tubes. As will be shown, it is a relatively simple affair to convert almost any tube preamplifier to meet the requirements of any of the strain gauge cartridges (even the Euphonics).

For those committed to the more conventional magnetic cartridge, it is necessary to consider the selected unit along with the associated preamplifier. The perfect magnetic cartridge must generate an electric analog of the groove modulations, but in accordance with the (highly artificial) RIAA curve. The associated preamplifier begins its work with a signal which must be changed to be of value. The development of competitive moving coil cartridges poses an additional problem... that of the sensitivity of the preamplifier. The vast majority of preamplifiers requires between 3.5 and 20 millivolts to produce adequate output, while moving coil cartridges produce about one-tenth of those figures: from 0.35 to 2.0 millivolts. There is a necessity for more stages, therefore, which compounds the problem.

Whether one uses a transformer, a circuit added to the basic preamplifier, or an additional (external) device, the so-called "head amp" or pre-preamplifier is subject to noise, distortion and other aberrations which are then amplified throughout the audio chain. Since tubes are inherently noisier than transistors, the most successful among these devices have been solid state. Though transformers are quiet, except for a decided potentiality for induced hum, they typically lack a bit—of dynamic range. One Nuvistor device, using a rare constant current circuit, is almost as quiet as its solid state competition, has excellent dynamic range, and provides enough gain to drive any preamplifier. Above all, it is musical' Though it is available from ADI and selected dealers, you'll find the schematic and construction details in this Manual.

The perfect preamplifier, then, cannot be properly described except in context: it is first necessary to state the cartridge type and output. The cartridge-preamplifier system, however, is electronically perfect if the output is an exact electrical analog of the groove modulations. Record wear and other practical considerations are important factors, of course, but at this writing it appears that the combination of Win Labs with a home-built tube power supply/preamplifier is potentially the most accurate, closest to theoretically perfect, transducer system available to date. The new Win is outstanding. Marketed in February 1977, the new Win box and linear cartridge is an overwhelming experience. The combination certainly approaches perfection.

Once the signal leaves the preamplifier, however, a new set of problems arises. Power amplifiers favored by today's audiophiles range from a $25 used Eico to a 3200-watt per channel solid state behemoth. With a pair of relatively efficient loudspeakers, the two units (as manufactured) are virtually indistinguishable at low listening levels. If the Eico has been properly modified, many will prefer it to the giant solid state unit for all but the more demanding tasks. How could this David successfully fling stones at its gargantuan competition?

The answer lies in the description of the "perfect" amplifier. The term "straight-wire-with-gain" provides the most easily understood explanation of the perfect power amplifier. It takes the signal from the preamplifier (or electronic crossover) and produces an exact copy at much higher voltages, with an internal impedance such as to permit application of this signal to loudspeakers. A number of reviewers, published in various audiophile magazines, have tried to create simulations of the straight-wire-with-gain situation in comparing the performance of amplifiers.

The term "straight wire bypass" has recently received undeserved acclaim ... undeserved because the author has learned that these models contain, of all things, potentiometers. One of the potentiometers actually used by a major publication was tested by the author and found to significantly modify the signal. During that test, the potentiometer was inserted in a circuit parallel with (literally) a straight wire. A common signal was passed through both legs of the circuit, and fed to a first-rate oscilloscope. Only the electronic difference was displayed; it should have been a straight line, of course. The wiggles and obvious aberrations

made it clear that any "straight wire bypass" cannot use that type of potentiometer, and makes use of any potentiometer questionable.

Even the most optimistic amplifier manufacturer must admit that his product does something to the signal besides amplify it. Whatever else happens can fall into several categories. The signal might not be amplified uniformly, which produces amplitude distortion. Sumanddifference signals might be generated and added to the original waveform; this is one cause of intermodulation distortion. Harmonic distortion, crossover distortion, transient intermodulation distortion (the newly infamous TIM), hum and noise will all be present to some degree or another. Each of these factors has an effect on the signal processed by the amplifier; each has a different effect upon the listener. Some are tolerable, even unnoticeable, at quite high levels; others produce a significant irritation in miniscule doses. The audible difference between the deluxe solid state amplifier and the old Eico may be attributed to the combination of detectable aberrations created by each; the net result from the Eico was simply less objectionable. This is usually attributable to the predominance of second harmonics, rather than high order harmonics, in the Eico's output, and to the fact that the output of a transistor is the logarithm of the input voltage, whereas the output of a tube is linear.

In defining the "perfect" amplifier, then, we will not attempt to relate it to the perceived sound, but will refer to that "straightwire-with-gain" concept which has sold so many amplifiers that do not even come close. In determining the relative merits and demerits of a specific product, it is worth the effort to apply the "straight-wirewith-gain" model to each of the performance parameters. This will permit an accurate prediction of (or explanation for) perceived sonic performance. Were one to participate in such a complex analysis, it would become immediately apparent that the ear-brain perception system can tolerate some types of distortion without discomfort. Low order harmonic and intermodulation distortion between one and five percent will probably be less unpleasant than TIM, or crossover ("notch") distortion at one-tenth the level. Hum and noise are a factor of limited importance; in any reasonably well-designed amplifier (tube or solid state) extraneous noise should be well below the troublesome level. The difference in perceived noise is insignificant as the S/N ratio goes from 100 dB to 70 dB at a normal listening level.

Dynamic range of an amplifier appears to be a major problem; a rough correlation can be made between that factor and rise time of the electronic chain. Further, it appears that the feedback circuit can (in any type of amplifier) improve "gross" performance and test data while eliminating some of the fleeting transients we depend upon in identifying music as "live". Amplitude distortion measuring techniques are not suited to detection of the equipment's capability to pass these transients; if they are present in the signal passed by the loudspeaker, though, your sensory system will know it.

Any reasonable modification program must optimize those technical performance parameters which contribute significantly to fooling the ear. This should be the case even if the resulting change in, say, transient response or intermodulation distortion is at the expense of harmonic distortion. If we begin the program agreeing that it is literally impossible to build a perfect amplifier, then it is obvious that we must do the best we can, accepting tradeoffs and compromises when necessary, but putting the emphasis upon those points which contribute most to "fooling" the sensory system. The perfect speaker works like the perfect cartridge in reverse. It converts an electrical signal to its mechanical analog, but then is faced with one additional task: it must couple that mechanical energy to the ambient air of the listening room. The transduction problem is difficult, but nearly perfect units have been built on a small scale; some of the headphones manufactured today are incredibly accurate. The coupling problem, with loudspeakers, can have a profound effect on the end product(perception), a result often independent of the effectiveness of tranaduction.

The perfect turntable is easy to define. It must rotate the record under the cartridge at a consistent and accurate speed, contributing no other relative movement between the cartridge and disc.

Since no turntable is capable of absolute consistency and accuracy, and (except for Gale) every commercial turntable has "rumble" at some level, is important to determine acceptable levels of performance. Because of potential compatibility problems, one must understand the relationship between specific faults or capabilities in one component with those in another. If the amplifier and preamplifier are solid state or are capable of unattenuated response at very low frequencies, any spurious subsonic signal generated by the turntable/arm/cartridge assembly may find its way to the loudspeaker. This may cause near-saturation of the amplifier, leaving insufficient margin for the audible portion of the signal. The sonic result can be a lack of clarity of the average signal, and premature clipping on transients. Most modern turntables generate little spurious noise, as evidenced by the rumble statistics furnished by the manufacturers. Unfortunately, the most modern of all can cause a severe problem. Rumble statistics are weighted; that is, a curve is imposed on the rumble figures which lessens the significance of subsonic signals. While the direct-drive turntable may have very low published rumble figures, a significant percentage of the rumble is "weighted out" because it lies in the subsonic region. If the cartridge (such as the Win, Panasonic or Stax) is capable o£ transduction at such frequencies, and the associated electronics will pass them, overall reproduction can suffer. Use of a rumble filter will help, but a simpler solution -which may have less effect on the resulting sound is to use capacitively-coupled vacuum tube electronics. Though some such devices will certainly pass subsonics with little attenuation, selection of the capacitor values can reduce the effect markedly.

Though we have not discussed the perfect tape recorder, the perfect tuner, the perfect crossover (electronic or passive), or other ancillary components, it is now relatively easy to define the optimum audio system.

Such a system will detect, pass, amplify and transduce a signal with virtually no change, whether the signal originates at a microphone, on a disc, on tape or from a tuner. The system will neither add nor detract from the original signal, and residual noise levels will be at a point where they do not interfere with perception of the program material. With a "perfect" audio system, the perceived sound will depend utterly upon the "software", or program material constituting the input. With "perfect" software, in disc or tape form, the net result will be a duplication of the original performance.

Since no perfect component is available, at some point the audiophile must determine just what constitutes an acceptable level of performance, and where compromises may be made. It is also necessary to decide which performance parameters are most important. The easiest way of doing this is to listen closely to a number of truly fine audio systems, all playing the same program source, and decide which seems closest to a "live" performance. The very best of these systems will likely consist of vacuum tube components, which is unfortunate for several reasons. First, the nature of such devices demands an extremely high price. Second, it is virtually impossible to correlate commonly published distortion and power figures with the subjective impression of "live". Third, when listening to any two really high-quality pieces of audio equipment, the differences are extremely small and can be detected only by two processes: the A-B comparison is one; lengthy listening periods is the other. Further, the A-B comparison must be done properly if the results are to be valid. Listen for differences; listen to one component for five or ten minutes, switch equipment and listen to the same material for about thirty seconds. Return to the original device for another five minutes, with new material, and repeat the process. Most stores like to flip rapidly between two components; the results of such a technique are usually deceptive.

If the serious listener concludes that the most realistic (to him) sound available is coming through vacuum tube components, but he simply cannot afford today's prices on such devices, this book is for him.