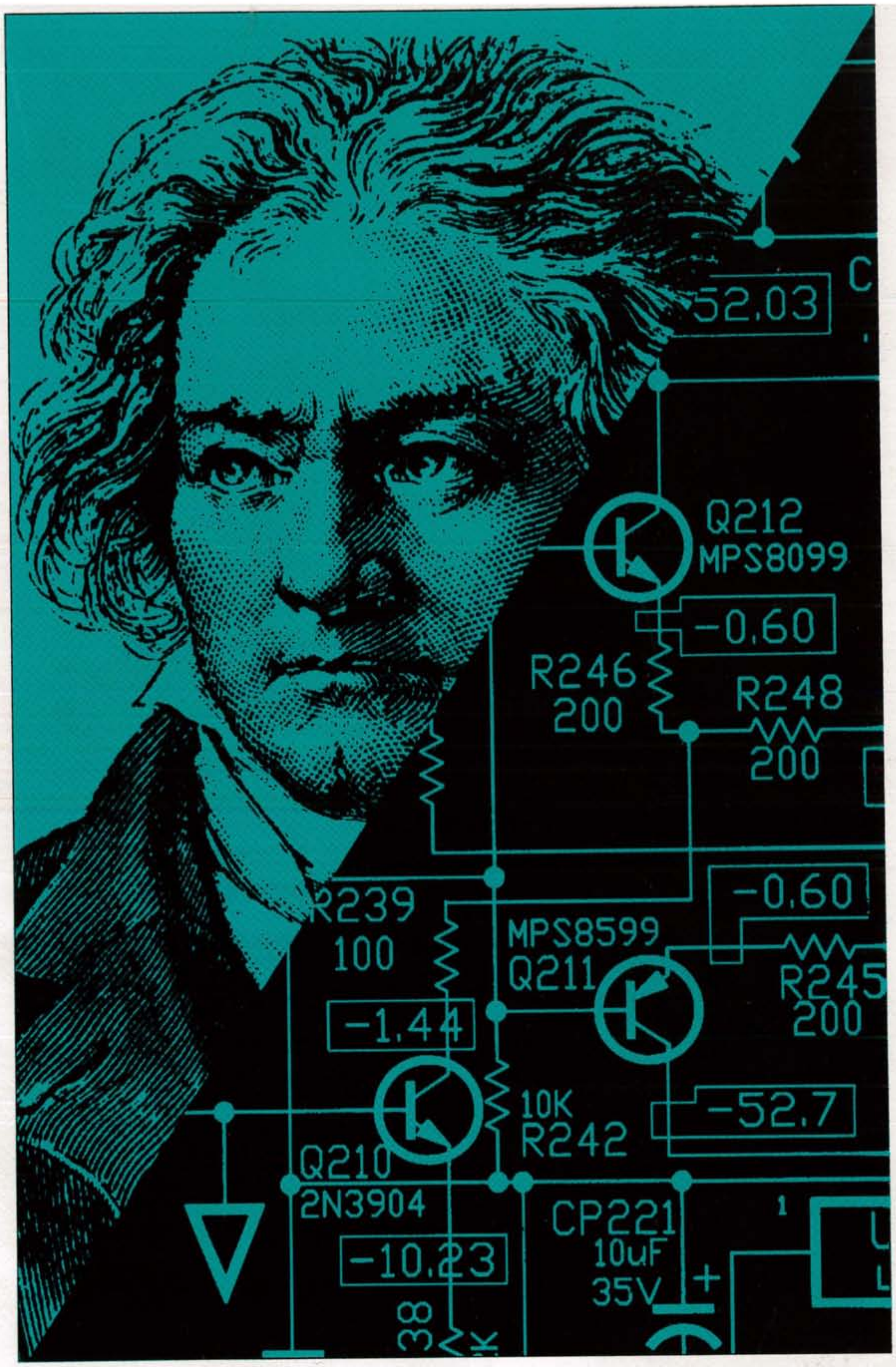
# THE ART & SCIENCE OF HARMAN KARDON




TRAINING GUIDE

harman/kardon

*Contents:*

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SECTION 1

# THE HARMAN KARDON HERITAGE

Consumer electronics is an industry with a strong focus on the here and now, and it's a little unusual for a company to dwell on its own past in a training guide. However, we're inviting you to ride on our particular time machine for some very good reasons.

First of all, Harman Kardon is one of the relatively few companies left in this business which has much of a past. Harman Kardon has produced more firsts and more significant technical advances than most other manufacturers. We've led the industry in product innovation from the early fifties right to the present. These accomplishments have built the Harman Kardon reputation, and we think you should know about them.

For Harman Kardon the past is more than simply a list of successful product introductions, it's a living tradition that continues to shape our corporate culture and our way of doing business. Our founder, Dr. Sidney Harman is still very active in our activities, and he provides us with a continuity of purpose and mission that are unique in our industry.

Harman Kardon has consistently offered meaningful improvements in basic audio performance, and we strive to make those improvements available at popular price points.

## EARLY YEARS — HARMAN KARDON AND THE GROWTH OF THE HIGH FIDELITY INDUSTRY

When Harman Kardon first started operations in 1953, the high fidelity industry was still very young. Phonographs of one sort or another had been sold to the public since the 1890s, but until the late forties most record players were low fidelity mechanical devices with horns for amplification, and very limited amplitude and frequency range. True, electric phonographs complete with amplifiers and loudspeakers had been developed and marketed in the late nineteen twenties, but the Great Depression and World War II had stifled the consumer electronics industry for more than fifteen years, and had hindered their acceptance. As late as 1950, the realistic reproduction of music in the home was still relatively novel.

Then, rather suddenly, after World War II, the country was ready for the emergence of high fidelity.

The component high fidelity industry which emerged in the early 1950s literally had its start in garages. Young engineers began selling kits to the public, as well as offering limited production hand wired components. At the same time a few companies began to offer loudspeakers, though for many years most listeners built their own based on published designs.

Gradually, manufactured items replaced the kits, but the majority of early components readily betrayed their kit ancestry through rough finish and utter lack of concessions to user convenience. The idea of high fidelity components for the nontechnically inclined music lover was still in the future.

## THE BIRTH OF THE RECEIVER

From its very beginning, Harman Kardon was dedicated to producing audiophile components, but Dr. Harman also wished to make practical, affordable products which would reach a wider market, while retaining the performance of the semi-custom products. In 1954 he introduced the very first receiver, incorporating an amplifier, preamplifier and tuner all in one chassis — a design which offered performance comparable to that of high quality separates, but at a substantially lower price. That first Harman Kardon receiver brought high fidelity to hundreds of thousands of people, and literally revolutionized the audio marketplace.

Then, just four years later, Harman Kardon jolted the industry again by introducing the first stereo receiver. While other companies were telling the public to convert to stereo by purchasing two monophonic amplifiers, Harman Kardon showed a real commitment to stereo audio by designing true stereo components and making them available at reasonable prices. Harman Kardon's response to stereo once again established our credentials as the technology leader, and placed our company firmly in the forefront of one of the most far reaching transformations to occur within consumer audio.

## INNOVATIONS IN AMPLIFIER DESIGN — ULTRAWIDE BANDWIDTH

Harman Kardon has never been a company to rest on its laurels. In 1963, we again created a stir in the industry by bringing out the Citation 3, a component power amplifier that introduced one of our key design concepts: ultrawide bandwidth.

In 1963 vacuum, tubes were still used in the best amplifiers, and transformer limitations confined most of them to a 20kHz upper limit at best. The Citation 3 had full power bandwidth approaching 140kHz, representing an amazing feat of transformer engineering that has never been surpassed. At first competing designers scoffed, but music lovers quickly confirmed what our own listening experiments had long indicated. Wideband response makes for much more accurate reproduction of music. We subsequently turned to solid state devices, but we never abandoned the concept of wide bandwidth. And just as we'd made the best tube amps in the sixties, we proceeded to make the very best solid state amps in the seventies, eighties and nineties, by continuing to design for ultrawide bandwidth.

## THE FIRST HIGH FIDELITY CASSETTE RECORDERS

In 1971 Harman Kardon sparked another revolution when we took a very close look at the then lowly cassette deck — at that time primarily a device for office dictation. We felt the format itself had considerable potential, and we had become aware of some very promising work being performed by Dolby Laboratories, then a small professional audio manufacturer which developed the first successful noise reduction system for magnetic tape. When we learned that Dolby had developed a simplified version of their professional system which was specifically intended for cassette applications, we immediately designed and brought out the very first cassette decks to feature Dolby Type B. Again we succeeded in reading the future. Within five years almost every major manufacturer had adopted Dolby B, and the cassette medium had largely superseded open reel in consumer applications.

## PHASE LOCKED LOOP TUNING

The following year we revolutionized FM tuner design. By 1972 stereo FM was a very well-established format and had proven itself capable of excellent fidelity. Unfortunately, much of that fidelity was lost in the tuners and receivers which were then being sold to the public. Station drift was the biggest single limitation in reception at the time, so we set about developing a solution.

That solution involved a circuit called the phase locked loop which had long been in use in critical military applications, but which was then considered too difficult to design and implement in consumer products. A phase locked loop continuously compares the oscillator frequency with a reference — usually a quartz oscillator — and instantly corrects for drift. As well as ensuring stability, it also facilitates extremely high tuning accuracy because it readily allows the substitution of a digital microprocessor for mechanical controls. This all takes place with an adverse effect on audio quality.

The phase locked loop was fairly exotic technology at the time, and we raised a lot of eyebrows by putting it in a consumer product. Not surprisingly, within a few years of our introduction of phase locked loop tuning, the industry adopted the system universally for FM tuners, and today the system is also in wide use in televisions and satellite receivers.

## AMPLIFIERS AGAIN

In 1977 we completed a long term investigation of existing solid state amplifier design, and we concluded that some new directions were needed. Early solid state designers, for the most part, had become fascinated with the ease with which large values of overall feedback could be applied to solid state circuits. They concentrated on reducing harmonic distortion to the exclusion of all else. Designs based on such excessive and obsessive use of feedback tended to measure well on standard distortion tests but they generally exhibited a harsh, irritating sound quality that detractors dubbed "the transistor sound," even though it had nothing to do with transistors per se.

At Harman Kardon we had been using sophisticated arrangements of local feedback loops with overall loops in our last tube amps of the middle sixties, and we applied the technology to solid state. We reduced overall feedback to under 25dB — as compared to 60dB and more in competing products — and we noted substantial improvements in sound quality. Our design techniques were almost immediately copied by "esoteric" manufacturers, and low feedback became the audiophile standard.

Meanwhile we discovered another key element in improving overall sound quality, that is, providing the amplifier with a high current capability. A few premium priced class A designs by other manufacturers had previously achieved high current output as a byproduct of class A operation. We found from our research and listening experiments that high current itself was actually more significant than class A operation in determining sound quality.

At the same time we became aware of the work of Matti Otala, a Finnish professor of electrical engineering who demonstrated that ordinary loudspeakers imposed short term current demands that weren't being met by existing designs. Dr. Otalas's research confirmed our own findings, and eventually we hired him to assist us in designing a new generation of power amps with power supplies and output stages equal to the task of driving real world speaker loads. The Citation X introduced in 1980 was our first high current amp. Every subsequent amp we've produced, including the amplifier sections of our lowest powered receivers, has had exceptional current capability vis a vis its nominal power rating.

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## CASSETTE AGAIN: DOLBY HX AND HX PRO

The years to follow saw Harman Kardon stepping up the pace of innovation. In 1981 we introduced the world's first cassette deck to feature the Dolby HX dynamic biasing system which resets bias continuously based on signal level for increased high frequency dynamic range. When the improved HX Pro system was introduced shortly thereafter, we were one of the earliest manufacturers to adopt that, as well.

## ACTIVE TRACKING

In 1987 we introduced a major improvement in FM tuners which remains unique to our product line. This innovation is known as active tracking, and it utilizes an additional phase locked loop in the IF section which rejects adjacent channels by a margin of 70dB with no loss of stereo separation or increase in distortion. In crowded urban areas with strong adjacent channel interference, our active tracking tuners are unrivaled by competing tuners at any price.

## HIGH VOLTAGE/HIGH CURRENT AMPLIFIERS

In the same year we introduced our high voltage, high current power amplifiers which utilized new transformer technology to mate the amplifiers optimally to different loudspeaker loads. High current amplifiers impose considerable stress on their own power transformers when forced to operate continuously into low impedance loudspeaker loads. By making the power supply voltage to the output devices switchable, current flow through the outputs is reduced without damping or loss of dynamics. An amplifier with switchable power supply voltage will put out the same power into 4 ohms as into 8 ohm and with absolutely no increase in distortion. It's an extremely sensible way to design a power amplifier.

## IMPROVEMENTS IN COMPACT DISC PLAYBACK TECHNOLOGY

Meanwhile Harman Kardon continued to grow and diversify. In 1983 we entered the autosound market with a line of power amplifiers which immediately became the standard of that industry, and introduced what many regarded as the world's best moderately priced turntables. But through most of the mid and late nineteen eighties the then new CD format engaged much of our research energies.

CD had been touted as

perfect sound forever, but our ears told us there was room for improvement. We saw the same phenomenon occurring as had taken place earlier in the amplifier field — mass market manufacturers reciting standard specifications which had little to do with perceived sound quality.

Our first inclination was to manufacture our own discrete output amplifiers for our CD players in place of the cheap IC buffers used in competing players. There was an improvement in sound quality, but still we weren't satisfied. Our engineers reasoned that the high frequency switching noise inherent in digital circuits was probably leaking into analog output sections, and in 1988 we devised a way to keep it out entirely — a change coupled interface between the D/A converter and the output amplifier circuitry which passed only the signal while blocking noise.

In 1991 we achieved a further improvement in CD sound quality by adopting the new 3D bitstream system across most of our product line. Bitstream, described in detail in a later chapter, is a means of achieving sixteen bit accuracy with what is in essence a one bit decoder. Ordinary decoders, which may have bit rates as high as twenty, assign different values to each bit, and each bit must have its own critically tuned conversion circuit. Such decoders are very difficult to calibrate and tend to drift over time, whereas one bit decoders are inherently accurate and immune to drift. We were the first company to adopt bitstream across our line, and the first to integrate this type of decoder with optimal front end and output circuits. We were also the first to utilize this technology in a CD changer. Again we seemed to have anticipated the future, because other manufacturers have adopted the bitstream type of decoder. However, no one selling similarly priced products has seen fit to emulate us by combining bitstream with properly designed analog circuits and electromechanical devices.

## CASSETTE YET AGAIN

The improvement in sound quality made it critical that cassette technology improve, as well. In 1990 we saw the first significant innovation in cassette deck design in almost a decade, Dolby Labs' "Type S" noise and distortion reduction system. Dolby S is a multi-band system providing noise reduction across the entire audio spectrum, vastly increased high frequency headroom, and an utter absence of audible artifacts. Dolby S is so effective that even highly experienced listeners have trouble distinguishing a Dolby S copy made on a correctly aligned deck from a CD or mastertape original. Dolby S provides digital dynamic range within the existing cassette format, and we believe that it is the most accurate consumer recording format for the present.

## AUDIO + VIDEO = ENTERTAINMENT & ENJOYMENT

For most of the last fifty years, audio and video were considered two separate worlds. The audio world was primarily devoted to the recording and reproduction of music. No matter how the music was played back, whether it be from a vinyl record, a cassette, a tuner or CO, we listened to in two channels since the dawn of stereo in the 1950s. Audiophiles derided the sound associated with video as primitive and to be avoided.

The other world is Video. Once strictly the domain of console TV sets and over the air broadcasters, it has changed dramatically in the past fifteen years. More American homes have television sets than indoor plumbing, and most of those sets are color. Screen sizes range from tiny two inch battery operated sets to 35 inch giants, to large rear screen projectors and to front projection systems with ten foot screens. Program material once limited to three networks and a handful of independent stations has grown to a wide array of choices. New broadcast networks, local stations, cable, pay-TV networks, pay-per-view movies and events, direct to home digital satellite systems, VCR, laser disc, CD Video, video games of all descriptions and even computers all provide video and audio source material to your customers' homes.

With the arrival of the new video sources has come an increase in the quality of the audio that accompanies that video. The result of the merger of a wide variety of video sources with high quality video and audio signals is the explosive growth of home theater, and as you would expect, Harman Kardon is a leader and innovator in that market, as well.

One of the first things that we realized was that the dynamics of movie soundtracks, the predominant home theater source material, are no different than that of the music played by audiophiles for years. If anything, the dynamic range requirements of an action movie are greater than those of many musical works! It quickly became evident that home theater was not an excuse to sell underpowered products. Indeed, it created a greater need for the ultrawide bandwidth, high current amplifiers developed by Harman Kardon.

Adding to the list of Harman Kardon firsts is the results of the research work done by two companies which are now part of the Harman International family, Lexicon and Fosgate•Audionics. Both have long histories in developing state of the art products for home theater. They shared the honor of introducing the first two high end surround processors to receive Home THX certification.

Fosgate•Audionics, in particular, is now a part of Harman Kardon, and the Digital Steering Logic™ circuitry developed and patented by Jim Fosgate will become part of Harman Kardon products in 1995.

Home theater is not simply a buzz word for Harman Kardon, instead, it is a way for us to use our existing technology in new markets, and an opportunity for us to develop new, groundbreaking technology for the future. At Harman Kardon we see the popularity of the home theater market as a way to deliver added enjoyment and entertainment to our customers through the combination of audio and video.

## PAST AND FUTURE

Harman Kardon is proud of its place in the industry, and of the way in which our engineering work has led to better reproduction of recorded music and increased listening pleasure for our valued customers. We've seen the industry grow and mature, and we've seen the technology of sound reproduction change, but we've also seen our basic approach validated time and again from our earliest days to the present. Design high quality components which provide uncompromised musical performance while offering the highest value, and one will never lack for loyal and satisfied customers. We have followed this course in the past, and we will continue to follow it in the future. The remainder of this guide indicates how our design principles are applied throughout the current product line.

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SECTION 2

# THE SCIENCE OF MUSICAL SOUND

Yes, it's quite a mouthful, and note we said musical. The science of sound in general is called acoustics, but the science of musical sound doesn't really have a name. Yet it lies at the foundation of what we're doing.

As an audio equipment manufacturer, we're primarily concerned with those aspects of the science of musical sound having to do with the reproduction of music. Music reproduction, by formal definition, is the process of drawing information from a recording and attempting to reconstruct the precise pattern of soundwaves comprising the actual musical event that was recorded. The process of music reproduction is always imperfect since reproduction inevitably adds or subtracts to these patterns of soundwaves. Any change which occurs in the waveform of the audio signal during reproduction detracts from enjoyment of music and our ability to immerse ourselves in the performance, and we thus strive always to reduce such alterations to an absolute minimum.

In the sections that follow, we're going to talk about both the nature of sound and the way that representations of sound become distorted in the reproduction process. In particular we are going to provide extended definitions of sound waves, sound intensity, frequency, and phase. Later we will discuss the nature of distortion, and finally we will attempt to indicate what makes for accurate reproduction of sound.

**Sound Waves**

All sound, musical or otherwise, consists of a series of expanding pulses or pressure fronts caused by the vibratory movements of some physical object. The vibrating physical object producing the sound exhibits some kind of back and forth motion that alternately causes the object to push against the air adjacent to its surface, and then to draw away, sucking the air back with it. The movement of the air in relation to the object exciting it gives rise to the actual sound waves we experience.

The back and forth motion of the vibrating object by turns pressurizes the adjacent air mass, and then creates a partial vacuum within that same adjacent air mass. Since air is elastic or springy, the pressurized air has a tendency to rebound, which in turn pressurizes another mass of air next to it, which behaves in the same way, and so on outward through space. A partial vacuum will behave in a similar fashion, propagating itself by causing an adjacent volume of air to rush into the area of low pressure, which in turn creates another temporary vacuum in the area occupied by that adjacent volume of air, and so on. In this manner a sequence of moving high pressure/low pressure zones is propagated out into the listening space—generally in the form of an expanding sphere.

The result of one complete back and forth motion encompassing the formation of one high pressure mass followed by one low pressure mass is called a wave cycle. The wave cycle is commonly represented by an S curve passing above and below a horizontal axis line (Fig. 2a). The axis line is often called the zero crossing or zero reference level.

The propagation of sound necessarily involves movements of air molecules, but the individual molecules don't move very far. The motion is more like a stack of dominoes falling over. It's a matter of a transfer of energy across a space, not the movement of an individual air molecules across the entire listening space.

You can create wave cycles simply by fanning your hand in front of your face, and you will feel masses of air striking your skin in time with the movements of your hand (Fig. 2b). And you will have created sound, but you won't hear the sound, because the frequency of your hand movements is too low to create the sensation of sound. Only when the wave cycles occur sufficiently rapidly will the vibrations of the air come within the threshold of hearing.

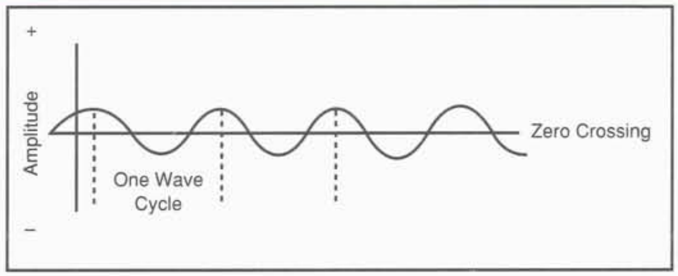
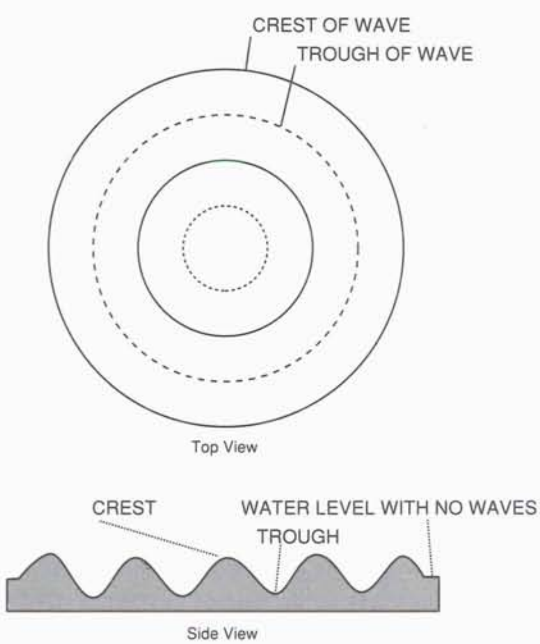


FIG.,2b

Representation of a wave form.



Waves created by a stone falling into a pool of water

FIG. 2a

## Frequency

If you've been interested in audio equipment for any length of time, you're undoubtedly familiar with the

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term frequency, but we'll review the definition anyway. Frequency refers to the number of times per second the vibratory movement or wave cycle repeats itself. Generally in technical writing on the subject of audio the word Hertz(Hz) is substituted for cycles per second, and thus the designation of 10 Hertz has precisely the same meaning as 10 wave cycles per second. In electronics in general engineers will frequently deal with wave cycles occurring at rates as high as several billion times per second, but in audio we're generally concerned only with frequency rates of under 14,000 times per second.

The concept of wave motion has, as you're probably aware, applications far beyond the sphere of acoustics, and the principles of wave theory are commonly invoked in discussions of optics, radioactivity, alternating electrical current, and electromagnetism. As it happens, the science of musical sound has to do with three of these physical phenomena—alternating electrical current and electromagnetism as well as acoustics.

## THE AUDIO FREQUENCY BAND

Vibrations in the air start to become audible to the human ear somewhere around twenty times per second, though the threshold frequency rate varies from individual to individual, and also with the intensity of the vibrations. On the high end, human auditory sensitivity starts to taper off above 10 kilohertz (ten thousand Hertz), and most adults cannot hear

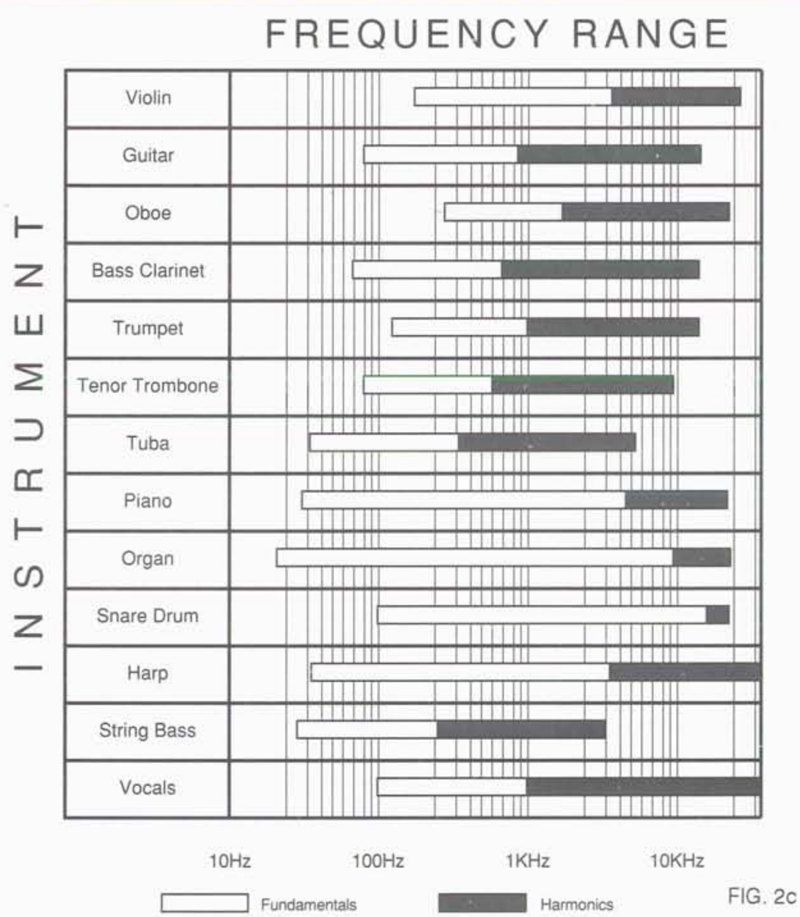
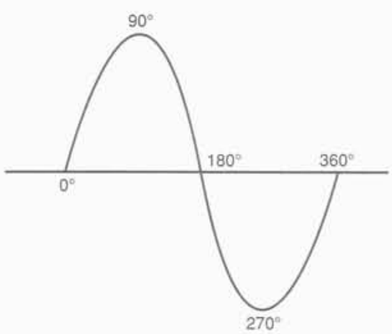



FIG. 2c

Frequency ranges of some common musical instruments fundamental tones above 15kHz, though children and young adults raised in quiet environments can hear out to twenty kilohertz and even a little beyond (Fig. 2c). Conventionally the audio spectrum is said to span 20Hz to 20kHz.




Sine Wave

FIG. 2d

**Sine Waves**

Modern acoustical theory has provided us with a surprising and extremely important concept about frequency which should be grasped by anyone interested in making a serious study of musical sound or audio electronics. The concept is as follows: all sounds consist of combinations of regular frequencies known as sine waves (Fig. 2d). A sine wave may be defined as a wave cycle occurring at one perfectly regular frequency rate—let us say fifty cycles per second—and never varying, never speeding up or slowing down. According to theory, any sound, however complex, can be broken down into combinations of simple sine waves. Thus sine waves are the atoms of acoustics, the fundamental elements.

What do sine waves themselves sound like? They're hard to describe because they don't occur in nature—it takes a synthesizer to create one. The closest instrumental sound to a pure sine wave is the tone of modern metal bodied end blown flute. In general humans find pure sine wave tones by themselves uninteresting and unnatural sounding.

**Intensity**

In addition to its frequency or combinations of frequencies, a sound can be described in terms of its intensity which actually refers to the air pressure in the wave fronts. Instead of pounds per square inch, acoustical air pressure or sound pressure is generally expressed in decibels—a complex logarhythmic measurement of intensity whose mathematics are too involved to explain here. The reason that the pounds per square inch measurement is not used is simply a matter of the values involved. Even a fairly loud sound raises the air pressure in a wave front by only about one one hundred thousandth of a pound. The absolute magnitudes are incredibly tiny.

Everyone in audio is acquainted with the term decibels, or dB for short, and most people use the term very loosely. In the strict sense, a decibel can only refer to the relationship between two intensities, and never to an absolute level, and thus one sound will be two dB greater than another but not two dB absolutely. When we use the term dB in the loose sense, we're usually referring to some level that is referenced to the lowest level of audible sound. That reference level is arbitrarily defined to be 0 dB, and it is set by convention at 0.0002 microbar—one microbar signifying the normal atmospheric pressure of fifteen pounds per square inch.

You are probably aware that each 3dB increase in intensity represents a doubling in the mechanical power required to produce it. Thus a sound which is 100dB over the reference level requires many millions of times the power to reproduce as the reference level. This has important implications in playback equipment, as we'll see.

**Phase**

Sound waves have one other dimension apart from frequency and intensity, and that is phase. Phase is a somewhat difficult notion to grasp, so please read the following section very carefully.

The term phase has many definitions in audio, of which the most basic and fundamental refers to a point within the wave cycle (Fig. 2e).

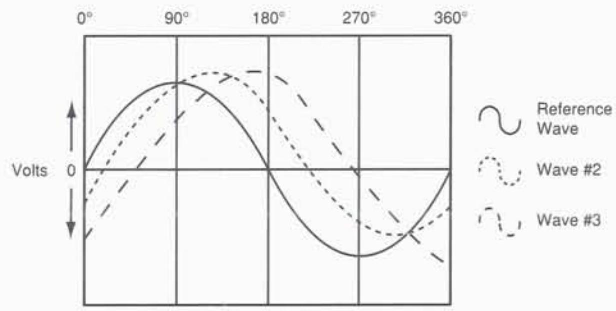



FIG. 2e

This requires some explanation.

As we have seen, the wave cycle encompasses the increase in pressure from a zero reference point to the peak of the wave cycle, then the decline in pressure to a low point an equal distance below the reference level, and then finally a return back up to the reference level—or, to put it another way, one wave peak and one wave trough. The familiar S curve used to describe this complete cycle spans a total of 360 degrees, the same number of points as are in a circle (Fig. 2f). Phase measurements refer to positions along this S shaped curve.

Now according to this definition of phase, the phase of a wave may be further described as its instantaneous intensity along the

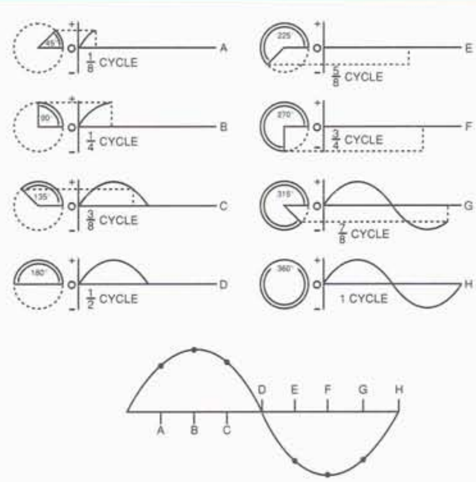



FIG. 2f

sinusoidal curve, since the S curve is basically a representation of varying intensity levels.

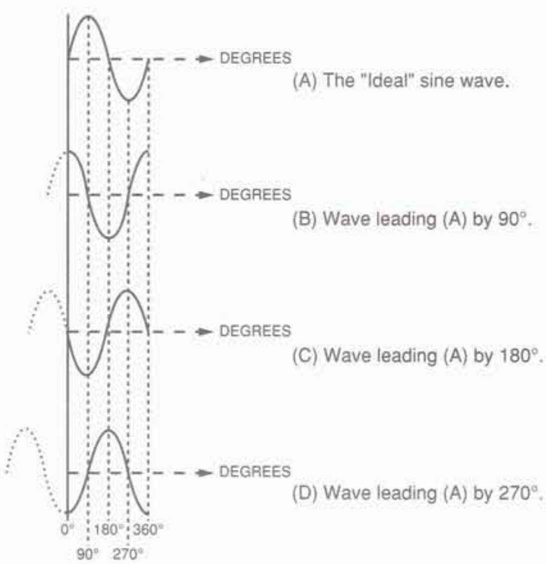
Now this instantaneous intensity we're discussing is always relative to single point in space since a wave has within its full length all intensity levels from peak intensity to least intensity. The phase of a wave only has meaning in terms of the wave impinging on a single defined point, and the intensity of the wave at that point relative to zero degrees of intensity determines its phase angle—that is, its phase measurement in degrees. To make this a little less abstract, let's consider the fixed reference points to be your eardrums. A sound wave coming at you from the side will strike each eardrum at a different point in the wave cycle, and thus the intensity level and the phase of the sound wave at each ear will be different.

If you think in terms of the wave moving past a single point you will see that the wave cycle must proceed through 90 degrees relative to a single point to reach its peak intensity, then another 90 degrees to return to the zero level, and yet another 90 degrees to reach least intensity. A full 360 degrees are required to complete the cycle.

To understand this notion of degrees of phase, it might help to visualize a straight line like the hand of a clock centered on the zero axis of the waveform just below the peak of the wave. The hand is moving clockwise, and when it's horizontal and facing to the left, the phase angle is zero. When it swings to a vertical position, the phase angle is 90 degrees. And when it sweeps back to the horizontal facing right the phase angle has increased a full 180 degrees.

Phase shift, another key concept in audio electronics, refers to the distance between two points, impinging upon a sinusoidal curve. Thus if a sine wave advances through one half of its cycle in relation to a fixed point so that the leading point of the wave is now one half cycle away from the fixed point, it has shifted its phase 180 degrees. By extension, the concept of phase shift can be used to define the relationship between two waveforms having the same sinusoidal curve. If both line up perfectly with one another, there are said to be zero degrees of phase shift between any two points on the waveforms because any one point will define the same relative intensity in either wave. If, on the other hand, the peaks of one wave completely fill the troughs of another, then the phase angle between the two is 180 degrees, in other words, one wave form has been advanced or shifted one half wave cycle in respect to the other. Of course two wave forms may have phase relationships spanning any other points in the wave cycles—thirty degrees, seventy-five degrees, one hundred and eighty-one degrees, whatever.

Phase relationships between different wave trains of the same frequency are important because they affect intensity. If, for example, you take two wave trains of the same frequency and superimpose one train on the other so that peaks line up with peaks and low pressure troughs with troughs, the pressure in each peak is doubled, while it's cut in half in each trough. The sound intensity of the two merged wave trains is therefore double that of each individually, and they are said to be in phase. But if you line them up out of phase so that peaks move into valleys, the pressure variations equalize, and the intensity of the merged waveform is reduced to zero, at least in theory. Such reinforcements and cancellations occur with reflected sound in a listening room, and they also take place in the electrical domain within amplifier feedback circuits and within crossover networks, among other places.




Demonstrating phase relationships of sine waves.

FIG. 2g

Frequency and intensity may apply to a single sine wave in isolation, but discussions of phase angles generally involve the relationship between two wave trains of the same frequency, referring specifically to how the high pressure peaks and low pressure areas line up with one another.

Phase relationships expressed in terms of phase angles, refer, strictly speaking, to different points along a single sinusoidal curve representing but one frequency, but audio engineers sometimes use the term phase in another sense to refer to the timing of one frequency in relationship to another. Thus if a sound wave of one frequency is launched before a sound wave of another frequency, they are said to be out of phase with one another. Such inter-frequency phase **(Fig. 2g)** differences are also known as propagation delays, and specifically relate to the differences in the time intervals required for an audio reproducer to pass certain frequencies.

For instance, a loudspeaker might pass 40Hz several milliseconds after it has passed 400Hz even though both frequencies have entered through the speaker's input terminals at the same time. In that case we would say that 40Hz has been subjected to a phase lag or time delay in relation to the higher frequency, though the actual phase angle represented by a given timespan will differ according to frequency.

We might add that phase differences in terms of points in the wave cycle are precisely related to time differences, and with each frequency the time value for a specific phase shift is different. For instance a 180 degree phase shift at 10Hz involves a delay of 500 milliseconds. The same phase shift at 20Hz only occupies 250 milliseconds. That's because a wave cycle at 10Hz has twice the time span of a wave cycle at 20Hz.

Perhaps the most important thing to remember about phase is that the term is used in several ways, and that a number of meanings may be expressed within a single discussion. Generally in discussions of equipment performance we are most concerned with phase as time relationships among different frequencies.

## WAVES INTO MUSIC

So what's all this have to do with music and musical reproduction?

Remember our discussion of fundamentals earlier in this chapter? Let's go back to that for a moment.

A fundamental is a sine wave of a given frequency which is sufficiently intense to give the sound a definite perceived pitch. For instance the key of Middle C on the piano produces a strong sine wave of 261.63Hz. A prominent fundamental is one of the characteristics that distinguishes a musical sound from noise.

Now the Middle C piano string also produces sine waves of other frequencies simultaneously, and these are called harmonics or overtones. (Somewhat confusingly, the fundamental is termed the first harmonic, while the first overtone is called the second harmonic). The harmonics give the individual musical instrument its distinctive character.

Harmonics are **(Fig. 2h)** conventionally assumed to be spaced at multiples of the fundamental frequency. For instance, the second harmonic (or first overtone) of Middle C is 523.25Hz—one octave up or twice the frequency (a musical octave is the interval between two notes in a 2:1 frequency relationship, not a set number of frequencies). The third harmonic of Middle C is roughly 783.99Hz, up the interval of a fifth. And so on.

This in theory. In reality the overtones of real acoustical

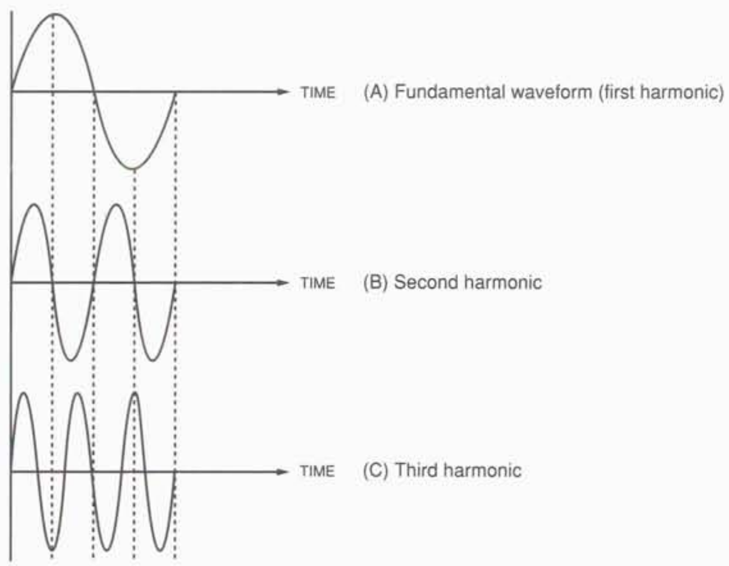



FIG. 2h

instruments don't strictly follow the harmonic series. They're shifted in pitch from where the true harmonics would be, and shift varies from instrument to instrument. This overtone pitch variation serves to distinguish the sounds of different instruments, that is, the timbre of instruments, to use the musician's term.

All acoustical musical instruments produce overtones, but the relative strength of the overtones varies according to the type of instrument. A modern end blown metal flute produces very weak overtones, and the fundamental is mostly what you hear. A violin produces a strong second harmonic and diminishing values of third and fourth, but the higher harmonics remain audible out to the tenth in some cases. A trumpet produces very strong high order harmonics, and not much second or third. (Remember we're using the term harmonic loosely here; harmonic is a mathematical term, and overtone is more correct when referring to the sound of actual musical instruments.)

Timbre involves more than just the overtone series however. It also involves a mixture of nonmusical noises that are a function of the mechanical operation of the instrument. In wind instruments, for example, the turbulence of the air in the pipes gives rise to various noises. In a guitar, the initial impact of the plectrum on the string creates a complex noise unrelated to the pitch of the string. On a kettle drum the whack of the stick on the drumhead is spectrally very different than the sound of the resonating diaphragm. Since noises represent extremely complex combinations of sine waves with no clearly identifiable fundamental, the noise combined with the musical tone can literally fill the audio spectrum.

So one instrument playing

one note can produce a lot of different sine waves at once. One instrument playing chords produces even more. Several instruments playing at once can produce hundreds or even thousands of different sine waves simultaneously.

All of these hundreds or thousands of frequencies occurring together create single complex waveforms at the eardrums of the listener or at the diaphragm of the microphone being used to record the sounds. If the microphone is mechanically perfect, and some made today are pretty close to perfect, the recording device will receive an electrical signal off the microphone preamp which theoretically contains all of the information needed to recreate the sounds which occurred in the microphone space, though one microphone by itself won't capture the spatial cues perceptible to a pair of human ears..

The only trouble is that a lot of electronic devices come between the microphone output and the loudspeakers in your listening room.

And that's where we come in, because we manufacture the electronics and the speakers for the playback half of sound reproduction chain.

## ACOUSTICAL SCIENCE AND THE PLAYBACK CHAIN

For playback to be considered perfect, the playback equipment has to preserve the exact shape of the complex waveform coming out of the microphone (we're assuming the distortions in the recording medium are fairly negligible). All of the separate sine waves must reproduced at the exact same frequency as when recorded, and must be at the correct relative intensities, and must also remain in the same time relationships as they were initially. In addition, no new frequencies must be added.

Modern audio equipment is generally pretty good about preserving the pitch of the recorded frequencies. If a 400Hz frequency was recorded, it's usually going to stay at 400Hz through the signal chain, though some cassette decks do have problems in this area.

Intensity differences tend to be less well preserved. Some equipment has trouble handling extremes of level.

Phase and time relationships are the least well preserved relationships, generally, and most audio equipment cannot maintain signal integrity in this area. Loudspeakers usually have poor phase integrity, with some notable exceptions, and so do some amplifiers, typically those with limited bandwidth. Some CD players have very poor phase response in the top octave as well.

Many authorities believe that flat phase response is essential for preserving the transient attack characteristics of instruments, and for rendering a convincing stereo image. In recordings of acoustical music done with simple stereo microphone arrangements, the mikes will pick up room reverberation in precise phase relationships to the direct sound, and such reverberation provides the ear with information on size and character of the listening space. If the audio system seriously muddles phase relationship, the ear simply isn't going to be able to unravel them, and the sense of the acoustic space will be compromised.

Probably the biggest failing of electronic components are the addition of frequencies not present in the recording—what we term distortion. Distortion may equal or exceed the levels of the recorded overtones for various instruments, and it can destroy the texture of a musical performance. And what's worse, the more complex the music, the worse the problem with distortion is apt to be (Fig. 2i).

All distortions are not of equal importance however.

The kind of distortion generally indicated in standard specification is harmonic

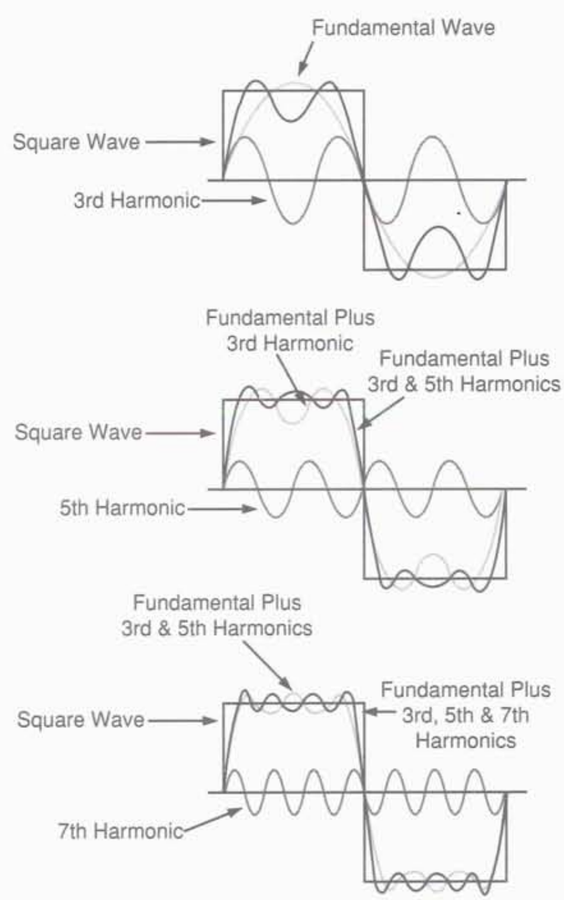



FIG. 2i

distortion, where the distortion takes the form of an added harmonic or multiple of a sine wave. When harmonic distortion is confined to the second and third harmonics, especially the second, it isn't very noticeable unless it is present in values of over 1%, but the higher harmonics tend to be very noticeable and very objectionable, even at very low values. Poorly designed solid state equipment tends to be characterized by low values of second and third harmonic but increasingly large amounts of higher harmonics, precisely the opposite of a desirable harmonic spectrum. Primarily because of the way specifications are measured.

Another type of distortion is known as intermodulation distortion which only occurs from the mutual effects of two or more tones reproduced together. Intermodulation distortions are what are known as sum and difference tones—one tone at a frequency that is intermediate between two tones that are intermodulating, and a second tone that is their sum. Intermodulation distortion adds a harsh, gritty quality to sound reproduction, and is particularly apt to be excited under transient conditions with certain designs of high feedback amplifier. In such cases it is known as transient intermodulation distortion.

There are other types of distortion as well, such as subharmonic distortion and enharmonic distortion, but harmonic and intermodulation distortion are the most significant.

We'll return to the subjects of phase and distortion at some length in the next chapter, because we strongly believe that the industry at large has tended to misrepresent the nature of distortion and phase, and to support methods of specifying performance that are extremely misleading to the lay consumer.

**To Repeat**

The object of an audio playback system is to reproduce a waveform with as little alteration as possible. Musical wave forms are extraordinarily complex, but can be reduced to combinations of simple sine waves of a single frequency. Apart from frequency, sound waves must be considered in terms of intensity, and their phase relationships with one another. A wave or combination of waves can be completely characterized in terms of frequency, intensity, and phase.

A musical playback system must preserve all frequency, intensity, and phase information perfectly and must add nothing additional—in other words, no distortion.

No audio playback system is perfect, but some imperfections are more tolerable to the human ear than others.

Modern audio equipment of advanced design can provide the listener with involving musical experiences. That's the kind we make.

3

SECTION 3

# THE FOUR DESIGN PHILOSOPHIES

Before beginning this chapter, make sure you've thoroughly absorbed the concepts we've discussed in the preceding chapter which dealt with the nature of musical sound and the requirements for accurately reproducing it. This chapter proceeds from Chapter Three and deals with the specific design criteria in electronic components which we have found to be necessary for the accurate reproduction of audio signals.

## THE FOUR IN BRIEF

All of the engineering criteria mentioned before ultimately have their expression in Harman Kardon's four basic design philosophies, namely a.) use 100% discrete components in the analog audio signal path, b.) design for ultrawide bandwidth, c.) use low negative feedback, and d.) provide high current capabilities. We think of these as the golden rules of audio engineering, and we'll have much more to say about each of them during the course of this chapter. We'd also like to point out that Harman Kardon is the only manufacturer in mainstream audio adhering to these principles.

But first a few fundamentals on audio circuitry.

## THE NATURE OF MUSICAL SOUNDING CIRCUITRY

At Harman Kardon we concentrate on electronic audio components. Some of these components are amplifiers, preamplifiers, tuners, receivers, cassette decks, and compact disc players.

The basis of all analog electronic circuits—the kind that still predominate in audio components—is the single stage amplifier consisting of one or more transistors, and one or more power supplies to energize the transistors—that is to provide them with the stable reservoir of electrical energy which they require to perform their functions.

Amplifiers get their name from their original purpose which was to increase the magnitude of an electrical signal without altering its waveform. Amplifiers initially made it possible to boost fading telephone signals and transmit them thousands of miles rather than a couple of hundred. Amplifiers also made possible the development of microphones and loudspeakers to increase the carrying power of the human voice and musical instruments. In addition, amplifiers played a key role in the emergence of the radio receiver where radio signals measuring mere billionthes of a watt could be boosted into audio signals of several—even dozens of watts of power. And of course ultimately, beginning at the end of the decade of the nineteen twenties, amplifiers were incorporated in the earliest high fidelity phonograph systems.

Later amplifier circuits would find other uses than simply boosting a signal. Amplifiers would be used to control the flow of electrons from a power supply to another amplifier that actually passed the signal. Amplifiers would be used to match electrical impedances from one circuit to another in order to ensure optimal transfer of energy. And amplifiers would be used in all sorts of signal processing functions requiring the dynamic control of various voltages within the circuit. In any place in the circuit where really precise control was needed, an amplifier of one sort or another was likely to be used.

All electronic audio components today contain multiple amplifiers. CD players, tuners, cassette decks, equalizers, surround sound decoders—they all have a multitude of low level amplifiers, and the accuracy of these amplifier circuits has a big influence on the ultimate sound quality of these components. In power amplifiers the amplification circuitry is the be all and end all.

At Harman Kardon we manufacture in their entirety all of the amplifiers used to pass audio signals in each and every product we make. We assemble those amplifying circuits piece by piece out of discrete, hand selected transistors, resistors, and capacitors. We are the only manufacturer of moderately priced audio components who can still make that claim.

We go to the trouble of assembling our own amplifier circuits instead of buying them off the shelf in the form ICs or integrated circuits because we have very specific design requirements which must be met. Observing these requirements results in superior sound, while failing to observe them is apt to result in harsh, fatiguing sound quality and limited resolution of the musical details within a recording.

All of these basic requirements are addressed in the four design philosophies listed at the beginning of this chapter. In the following sections, we'll examine each of these philosophies in detail.

## A FORMAL STATEMENT OF THE BASIC FOUR

1. Discrete devices (transistors, capacitors and resistors) should be used exclusively in the analog signal path.

2. Circuits should exhibit bandwidth of several times the audio spectrum to include all harmonic ranges.

3. Low negative feedback should be used in multi-stage amplifiers.

4. Power amplifiers should be provided with continuous high current capabilities.

We've formulated these criteria rules because we've found that circuits designed according to them simply sound better. Now we'll explain why, and please read the following sections very carefully because they cover fundamentals of high performance audio circuit design.

## DISCRETE DEVICES, WHY USE THEM?

Discrete devices are physically separate electrical elements that are manufactured and sold as units, and then connected into circuits, generally on a printed circuit board made of glass epoxy. They're soldered by automated assemblers, and the circuits, hand tested—at least that's how we manufacture circuit boards at Harman Kardon. There's a lot of pride of workmanship that goes into discrete audio components.

Integrated circuits, on the other hand are single chips of silicon on which transistors are formed, and on which resistors and capacitors are bonded. Most are mass produced, and most are purchased off-the-shelf from chip manufacturers by the mass market audio companies.

Discrete circuits and integrated circuits both use the same basic types of electrical elements—transistors, capacitors, resistors, diodes, and occasionally inductors—and in theory the same circuit configurations can be etched onto the silicon wafer as can be built on a circuit board. So why, you might ask, does Harman Kardon continue to favor hand assembled discrete circuits long after most of the industry has gone over to machine made ICs?

The explanation is a little involved, but it's worth following because it embodies what is really the whole rationale behind perfectionist audio electronics.

Given the state of today's manufacturing technology, an IC cannot be made on a cost effective basis that will challenge the performance of a competently designed, multi-stage discrete circuit in exacting audio applications. The reason for this can be found ultimately in the tolerances of the resistors, capacitors, and transistors, and especially the transistors. High performance circuits require parts of fairly tight tolerances, and to get such tolerances the parts must be carefully selected from a production run. On an IC where everything is made at once, this isn't possible. If one element on the chip is out of spec, you've either got to live with it, or discard the whole chip. Accordingly, manufacturing chips with high tolerance individual elements necessarily involves a high reject rate, and thus very high cost. And because making trim adjustments is very difficult on a chip compared to in a discrete circuit, the tolerances of the elements on the chip must actually be much higher for a given level of performance within the same circuit topology.

The huge majority of ICs

manufactured for audio use do not contain high tolerance elements or trim controls, and instead rely on certain types of circuit topologies or types which are tolerant of poor component quality. The differential op amp is one such topology, and it is very commonly used in audio ICs. Practically all such popular IC topologies are characterized by very high voltage gain, generally achieved by linking up a long chain of transistor gain stages, and then utilizing very heavy global feedback to reduce the gain to a manageable value. At the same time, feedback will be used in an attempt to reduce the distortion caused by all of the basic nonlinearities and imbalances within the circuit.

What's wrong with this basic approach? Nothing, if you accept the pronouncements of certain reviewers in the mass circulation audio press. And in fact even the cheapest ICs will usually exhibit vanishingly low static distortion, often outperforming discrete circuits in this regard. Nevertheless, ICs continue to be rejected by designers of perfectionist audio equipment for both consumer and professional applications.

ICs made according to the usual practice with low tolerance circuit elements, multiple gain stages, high voltage gain prior to the application of feedback, and high negative feedback, generally suffer from limited bandwidth, poor transient stability, and high transient distortion, all of which failings are closely interrelated.

In addition, ICs suffer from one other major shortcoming which is not a function of circuit topology and thus is not susceptible to correction by substituting improved topologies, and exerting more stringent quality control. This limitation is absolutely fundamental to the devices and isn't likely to be overcome in the future. It's a matter of simple heat dissipation.

Transistors, whether discrete devices or components on an IC must be operated within a fairly narrow range of temperatures. Outside that range they're subject to nonlinearity and catastrophic failure. High temperatures are especially to be avoided, and in a discrete circuit they can easily be avoided by spacing the devices on the circuit board and supplying each of the high power transistors with a separate heat sink. There's just no way of doing these things on a chip, and for that reason, high power, all-IC amplifiers just don't work very well, especially not when the entire circuit is on a single chip. Accordingly, the single IC power amp is apt to be confined to low powered receivers, while units of over 50 watts will tend to have discrete output circuitry and everything else on a single chip.

For every one of these reasons Harman Kardon has scrupulously avoided ICs in amplifying circuits.

By this time you may be asking yourself why other manufacturers of popularly priced equipment favor IC circuits if their performance is so objectionable? Very simply because they are cheap and because most manufacturers are selling spec, not sound quality. IC amplifiers will produce low distortion outputs in certain standard test situations (testing of amplifiers has become a minor scandal in the consumer audio industry) while failing to perform adequately when attempting to produce music from real world loudspeakers.

## WIDE BANDWIDTH VERSUS LOW BANDWIDTH

The bandwidth limitations of commonly used audio ICs arise from several characteristics of these devices. The first has to do with the quality of the transistors themselves. Wide band transistors are difficult and expensive to produce, and, not surprisingly, the transistors etched onto the silicon chip that forms the low cost IC are incapable of wide band response. Moreover, the chain of gain stages within the IC forms a complex low pass electrical filter which further reduces bandwidth. In some ICs the high frequency response of the circuit falls off below 100Hz with the feedback loop open.

And to a large extent, the limited open loop bandwidth of such IC is entirely deliberate, in fact a stated design goal. That's because any negative feedback amplifier must attenuate the feedback in the high frequencies before the phase shift inherent in the transistor itself turns the feedback positive and causes the amplifier to oscillate, that is to produce a continuous and potentially destructive high frequency output. The designer attenuates feedback by rolling off the gain of the gain stage itself, a process known as compensation. And the more gain and the more feedback involved, the farther down in frequency the rolloff must begin.

In contrast, a properly designed discrete circuit can utilize specially selected transistors and local feedback loops to extend the (Fig. 3a) bandwidth out to several hundred kilohertz for each stage.

So why is ultrawide band

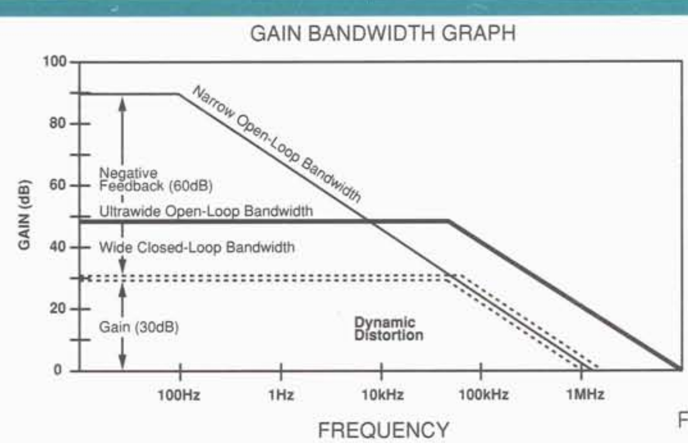



FIG. 3a

frequency response so important?

For a number of reasons, but before we discuss them let's define ultrawide bandwidth.

When we describe an amplifier circuit as being ultrawide band, what we're really saying is that the circuit maintains level response over a wide frequency range, and that the circuit is linear over that same bandwidth. A wide band amplifier for audio frequency use would have bandwidth of several times the audio range—minimally five times according to our standards.

Apologists for narrow band mass market designs are quick to object that no musical information is present beyond 20kHz in current digital formats, and that even the most extended analog recordings did not exceed 40 or 50 kilohertz at the top, and so therefore wide bandwidth is unwarranted. What they're ignoring are the effects of wide bandwidth within the audio spectrum even in the absence of ultrasonic information.

Ultrawide band amplifiers are far more phase linear than those with limited frequency response (Fig. 3b). Rolloffs in frequency response are always accompanied by phase shifts, and these can extend down well into the audio band. With a truly wide band amp the phase shift associated with the high frequency rolloff doesn't begin anywhere near the audio band.

And while program information may be absent above 20kHz, spurious electrical energy is not. If such energy excites the amplifier into clipping, oscillation, and distortion, then modulation products will occur within the audio band.

Furthermore, ultrawide band amplifiers will always exhibit better high frequency headroom than bandwidth limited designs, and thus they cannot easily be overloaded. Sounds like cymbal crashes, trumpet blasts, and bells will be rendered more convincingly.

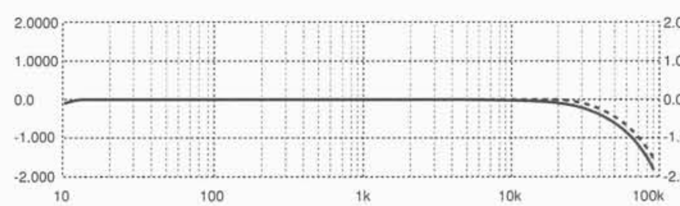



FIG. 3b

And this last point needs stressing. A low cost op amp is easily overloaded by high frequency signal content because the op amp's effective bandwidth prior to the application of feedback is so limited—usually far less than 1kHz. Furthermore, the filtering used to roll off the high frequencies within a gain stage will also slow down the feedback loop so that it cannot correct for transient distortion, all of which results in a double whammy—an increased tendency to produce distortion in the presence of high frequency inputs, and an utter inability to correct for such distortion when it does occur.

But the most compelling argument in favor of ultrawide band designs is that they simply sound better. During the early seventies when solid state amplifier design was coming of age, nearly all of the more progressive designers found by trial and error and a lot of listening that wide band amps sounded smoother, more open, and more detailed. Today ultrawide band circuits are quite prevalent in the high end, but in mainstream audio only Harman Kardon produces ultrawide band discrete amplifying circuits.

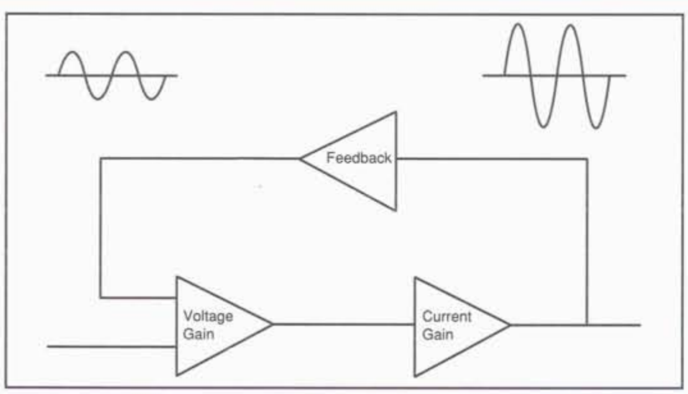
## THE ADVANTAGES OF LOW NEGATIVE FEEDBACK

Feedback has truly become a dirty word in high end audio. Many designers of premium priced esoteric audio gear proudly proclaim that their circuits use no negative feedback whatsoever. On the other hand, most engineering textbooks routinely prescribe high feedback for linearizing circuits, while many prominent journalists within the audio community assure us that high negative feedback is entirely benign. Who's right?

At Harman Kardon we make no secret of our allegiance to general principles of electronic design which have found wide acceptance among esoteric designers. We also believe for reasons stated below that the high feedback ICs predominating in affordable audio gear made today provide for less than satisfactory listening experiences.

But the issue isn't nearly so simple as the advertising brochures and buzzwords would suggest.

First of all, feedback of one sort or another is almost absolutely necessary in a functioning amplifier circuit. Indeed most amplifier circuits have many, many feedback loops and arrangements (Fig. 3c). Operating an amplifier without any feedback is an invitation to instability and really gross amounts of distortion. Blanket condemnations of




Block diagram of a power amplifier showing voltage gain stage, current gain stage, and the feedback loop.

FIG. 3c

feedback have no engineering validity. It's how feedback is used that determines its correctness, not its presence or absence.

When we talk about feedback in power amplifier circuits, which is usually the context of these discussions, we're mainly talking about negative voltage feedback, where a portion of the output of the amp is returned out of phase to emitter of the input transistor.

There are several reasons why this is done.

Negative voltage feedback lowers distortion. All active devices, whether tubes, bipolars or FETs, produce significant percentages of distortion during normal operation—often over ten per cent below clipping. That's audible, and totally unacceptable in a high fidelity circuit.

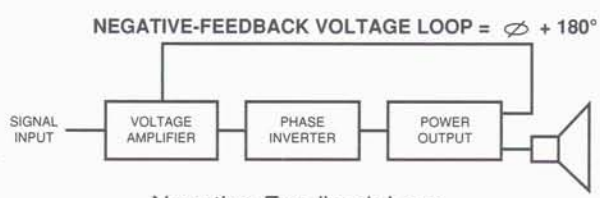
Negative voltage feedback also lowers output impedance for increased damping, constant voltage delivery, and flat frequency response in the output stages of an amplifier, and better drive capabilities in the preceding stages of the amplifier.

Negative feedback stabilizes the circuit. Any response irregularities in the output circuit of whatever nature will tend to be (Fig. 3d) canceled out by the feedback loop.

So feedback is not all bad, and savvy designers do not attempt to eliminate it entirely. What they do attempt is to design an amplifier circuit which is as stable and distortion free as possible prior to the application of feedback, and only then apply feedback to ensure even greater stability. And they also tend to rely far more on local feedback than on the global loop—local feedback being feedback around single stages and global feedback encompassing the entire circuit.

The designers of the penny ICs used by our competitors have very different priorities. In such a circuit topologies as they favor, the feedback loop assumes overwhelming importance, indeed high feedback is necessary in order for the circuits to work at all. Therefore the function of feedback in such a circuit is worth a close look.

It is one of the truisms of electronic engineering that high voltage gain is usually associated with high distortion as well. When high gain, high distortion stages are linked in a chain to get more gain, the distortion gets really serious because each succeeding stage




Negative-Feedback Loop

FIG. 3d

multiplies the distortion products of the preceding stage as well as adding its own.

When negative feedback is finally applied from the end of the chain back to the beginning, both the gain and the static distortion of the entire circuit is considerably reduced. The bandwidth of the circuit may also be increased if the feedback is applied selectively, that is, if the feedback is made very heavy in the low frequencies where gain is greatest, and then the feedback is progressively reduced with increasing frequency to match the drop in gain.

The use of feedback to reduce distortion is based on a very simple principle. The gain of the circuit may be increased much more rapidly than the percentage of distortion, so that the swap of gain for linearity through the feedback loop always appears to be advantageous.

The IC engineer is always willing to throw away gain to reduce distortion, and so theoretically he'd like to have infinite gain open loop. That's not really attainable, but some common ICs made today have open loop gain factors in the millions—at least at low frequencies.

All is not exactly what it seems, however. First of all, the feedback loop itself creates new distortions even as it is canceling out old ones. If, for example, the amplifier open loop creates a third harmonic initially, that harmonic will engender its own third harmonic when it's cycled through the circuit a second time via the feedback loop. A third of a third is a ninth, the ninth may itself engender a twenty-seventh and so on, though the poor high frequency response of the circuit eventually imposes a limit. At any rate, the high order distortions created by the feedback loop, though small in value, tend to be more audible than low order distortions.

Furthermore, the feedback loop will be entirely unable to cope with high frequency signals that pass through the circuit before the loop itself can close. Feedback is always after the fact, and when feedback is applied to multi-stage amplifier, the transit time of the feedback through the circuit can be a factor, especially when individual gain stages are compensated.

If the bandwidth of the gain stages were high as well as the gain, the designer could possibly get away with high feedback, but the bandwidth isn't high, and it can't be made high lest the amp become unstable. In a typical high feedback amp, everything is sacrificed to achieve low static distortion, that is to "make spec"—and sound quality suffers as a consequence.

The situation is worst when the signal going through the circuit is sufficiently intense to cause clipping. Feedback itself is a factor of gain, and when a circuit is clipping it runs out of gain. At clip a circuit is producing more and more distortion, and at the same time the feedback loop is less and less able to reduce that distortion, and at some point the circuit becomes essentially open loop. High feedback circuits always clip hard, that is, when they reach their limits and the feedback loop suddenly becomes ineffective, they react by showing their essential open loop character and generating very high distortion.

Remember what we said about IC's having very poor open loop response in the high frequencies? In fact they are very liable to clip when presented with intense high frequency transients, and the clipping produces an almost infinitely rich assortment of harmonics which is then modulated at each gain stage for a kind of distortion soup at output. The term for this kind of distortion is TIM (transient intermodulation distortion) a nomenclature which was coined by Dr. Matti Otala, a very noted theoretician of amplifier circuit design, who has long been a consultant for our company.

We believe that TIM lies at the root of the innumerable observations by discriminating listeners that IC circuits sound bad. Most IC amps produce high values of TIM when reproducing music, and exhaustive listening tests done in Europe more than a decade ago have pretty conclusively demonstrated TIM's audibility. On the other hand, Harman Kardon amplifiers produce no measurable TIM.

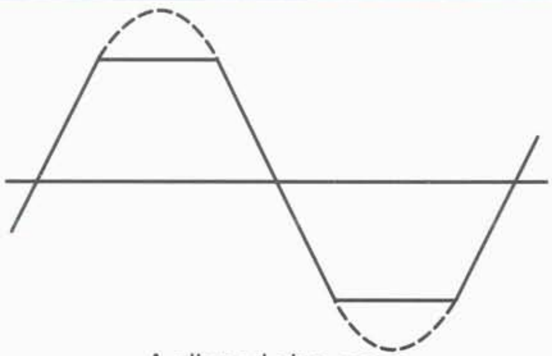
The companies manufacturing IC based components have, quite naturally opposed the formulation of industry standards for testing TIM, and have refused to publish figures on the performance of their products in terms of this particular type of distortion. And with very good reason. They can make a product that achieves low static distortion through high feedback, but they can't possibly reduce TIM through such means. In fact they can only make it worse.

## HIGH CURRENT CAPABILITY

High current capability as a design criterion is only applicable to power amplifiers. CD players, tuners, and cassette decks are never required to put out significant amounts of electrical current. It so happens that all Harman Kardon power amplifier circuits including those in our receivers are designed to put out high continuous current in relation to their nominal rated power.

To understand the importance of high current, we have to take a look at what an audio power amplifier does in relation to a loudspeaker.

In a physical sense an amplifier performs work, it doesn't simply transmit information. It provides a source of alternating current for an AC motor, that motor



A clipped sine wave.
Clipping this severe rarely occurs during actual music reproduction, and a waveform this distorted would destroy most speakers in seconds.

FIG. 3e

being the loudspeaker itself. The fluctuations in current comprise the audio signal.

The loudspeaker itself makes physical movements in order to energize the air in the listening space, and the force of those movements is a function of the electrical current presented to the speaker by the amplifier.

Wattage, the electrical measurement which is specified for amplifiers in standard ratings, consists of current times voltage, or amperes times volts. Loudspeakers in general, with the exceptions of the piezo type, which isn't used in high fidelity, and the electrostatic, which is rare and costly, are current devices—that is their output is proportional to current flow through the voice circuit not to voltage. Voltage is only important for overcoming the resistance of the wire and the voice coil, and ensuring current flow. Actual current delivery is what really counts.

Most solid state amplifiers made today are more or less constant voltage devices which will maintain constant signal voltage levels in the face of widely varying loudspeaker impedances, and will vary current flow according to the fluctuating impedance at the loudspeaker terminals—but only up to a point. Their ability to do so is determined by their current capability which in turn is a function of power supply engineering and storage capacity. (Fig. 3e).

Interestingly this fact is ignored in the standardized testing of amplifier power. Amplifiers are tested into a stable, precise eight ohm resistor which doesn't closely resemble a loudspeaker impedance at all. The eight ohm resistor makes very slight demands for current, and with a 40 volt output working into it, will permit the generation of 100 watts of power with less than three amperes of current.

Now loudspeakers, by and large, are inefficient devices with numerous electrical and mechanical reactances in addition to the DC resistance of the voice coil circuit, and these reactances alternately interfere with current delivery or short the outputs of the amplifier thereby making tremendous demands on the power supply. The feedback tells the amplifier to attempt to maintain constant voltage and to make such adjustments in current flow as are necessary to do so.

And here's where things get interesting. The adjustments in current flow necessary to overcome the mass of a woofer, return that woofer's own electrical kickback to ground, and to deal with momentary impedance nulls, may require the amplifier to deliver six times the current on a transient basis as is required by a perfectly resistive eight ohm load. Producing a 100 watt output into garden variety 8 ohm dynamic loudspeaker might very well require an instantaneous current capability of 20 amperes. An especially reactive 4 ohm loudspeaker requiring a 200 watt output could impose current requirements on an amplifier in excess of 50 amperes. In fact when we designed our cost is no object Citation XX back in 1981 we specified a 100 ampere peak output because we found loudspeaker, which would demand that much at high output levels. And today even in our lowest price receivers we engineer current capabilities which are literally multiples of those for competing products.

We will have much more to say on the subject of high current in the following chapter which deals specifically with the subject of power amplifiers, but we'll note that the inability of the amplifier to respond to momentary current demands is not without sonic consequences.

If the amp can't supply the current, it can't properly damp a woofer, and it often can't maintain level frequency response across the audio spectrum. And because the low current amplifier will not be able to maintain constant voltage either, its output waveform will be distorted.

In fact most inexpensive amplifiers are prevented from even trying to supply large amounts of current by protection networks which limit the underrated output circuits just to keep them from overheating.

It has often been noted by most listeners that Harman Kardon's lowest powered receivers sound more dynamic and powerful than competing units with two and three times the rated power, this is because the competing units commonly have less than half the current capability. As with cubic inches in an automobile engine, there is no substitute for current capability in an amplifier.

At this point you're probably asking yourself why everyone else isn't delivering high current if it's so very important. As a matter of fact the high end manufacturers of limited production audio amplifiers generally do design their equipment for high current operation, but the manufacturers of mass produced, moderately priced equipment do not. High current capability is a function of power supply design, and requires large value capacitors and high power transformers, both of which are bulky and expensive. And even with an adequate power supply, high current cannot be had with an IC output stage. Frankly we choose to put our manufacturing cost in our power supplies and in our signal circuitry while our competitors prefer to buy circuitry for front panel displays and "convenience functions". Life presents all of us with choices, and we have chosen to build components that are good sounding rather than ostentatious in appearance.

The Big Four in Summary

We can see from our discussion that discrete circuitry, ultrawide bandwidth, low negative feedback, and high current capability are not only necessary for accurate reproduction of music, but are entirely complementary design goals.

Harman Kardon played a pioneering role in establishing the importance of these four principles, and we have followed all of them in every product we've made since their inception.

In the high end these principles have come to be generally accepted, and we are always pleased to see equipment priced at multiples of our own exhibiting very similar design work. Imitation really is the sincerest form of flattery. We are equally gratified that no one in the mass market appears to have gotten the message. As long as they continue to rely on bandwidth limited, low current, high feedback ICs, music lovers on a budget will have no choice but to come to us.

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SECTION 4

# AMPLIFIERS

The preceding chapter on design philosophies provides a lot of the basics on our approach to both low level and power amplification. Here we examine some of the details of amplifier engineering. Specifically we will be concerned with the distinctions between voltage and current amplification, the ways in which amplifier circuits are utilized in various parts of the signal chain, and in the ways that the four basic design principles are embodied within each type of amplifying circuit.

## VOLTAGE AND CURRENT

Remember what we said about the single stage amplifier being the basic building block of analog audio circuitry? Let's expand on that a little.

Single stages of amplification can themselves be divided into three functional groupings, voltage amplifiers, current amplifiers, and buffer amplifiers. The first type increases the magnitude of the voltage variations in the audio signal, while the second increases the volume of current in the signal. A buffer amplifier, the third type, is ordinarily a type of current amplifier, but the current amplification factor is very low, and the purpose of the circuit is to optimize impedances between the preceding and succeeding stages. Buffers are most commonly used at the inputs and outputs of components, but they may occur internally as well. Ordinarily in the audio signal chain from signal source to the loudspeakers, there are several stages of voltage amplification, and a multitude of buffers, but only one current amplifier—that occurring in the output stage of the power amplifier.

## COMPONENT AMPLIFIER TYPES

Component amplifiers—that is those products containing amplifying circuitry and nothing else—take three forms. These are the so-called preamplifier, the power amplifier, and finally the integrated amplifier which is a combination of the first two. A receiver combines an integrated amplifier with a stereo tuner. At Harman Kardon we make all four types of components.

Component preamplifiers appear to be a product category in the process of being reconstituted. Traditionally they have usually included a phono amp, also called a phono section intended specifically for handling the outputs of phono cartridges, and a line amp or line section which accepts inputs from all other signal sources including the output of the phono section. Phono amps have heavy voltage gain, with gain factors in the thousands. On the other hand, the line section of a preamplifier generally has comparatively little voltage gain, and serves primarily as a buffer. In the future premplifiers will be increasingly likely to dispense with the phono section and possibly to include various types of signal processing such as surround sound decoding.

Power amplifiers themselves usually have multiple voltage amplifiers and one current amplifier. Sometimes a circuit will include an internal buffer amp before the output stage.

We devote considerable attention to the design of both preamplifier and power amplifier circuits, and we have developed different design techniques for each.

## PHONO AMPS

The phono stages of all Harman Kardon preamplifiers and preamplifier sections are distinguished by the use of passive equalization for the RIAA network. Ours are the only phono stages in any reasonably priced component to be so configured, though passive equalization is quite prevalent in perfectionist component preamps costing thousands of dollars.

For those of you who've gotten involved in the audio scene subsequent to the decline of the analog phonograph record, let us point out that phonograph records themselves are cut with a heavy emphasis in the treble and de-emphasis in the bass in order to optimize signal-to-noise ratios and avoid overcutting the record grooves, and thus the output of the cartridge must be heavily equalized in the phono stage to obtain a flat response.

We won't spend a lot of time on this area of phono amplification, except to say that phono stages are tricky to design, and that we're probably the only mainstream audio company making a phono amp with sound quality comparable to that of the better esoteric units—in large part because we use passive equalization combined with discrete circuitry. Most phono amps are made according to general "op" amp principles with extremely high open loop gain and heavy feedback, with equalization being done through the feedback loop. The high gain is unavoidable in a phono amp, but the heavy feedback need not be, because instead of using a feedback loop, one can flatten response with a passive network of resistors and capacitors (the term passive refers to a circuit without gain devices such as tubes, FETs, or bipolar transistors).

Why passive? Feedback equalizers have all of the undesirable attributes of feedback loops used for distortion reduction. They create modulation products, and they tilt the harmonic spectrum toward a preponderance of offensive high order distortions. Passive equalization generally sounds much better, and we are not the only manufacturer to have reached this conclusion.

## VOLTAGE AMPLIFICATION

In voltage amplification stages the desirable attributes are low noise and immunity to clipping under any conceivable signal conditions, and here is where discrete circuitry becomes vitally important. Many ICs will clip hard and audibly in the presence of voltage swings of less than two volts. The result is a scratchy sound on musical peaks, and a noticeable loss of resolution generally. In contrast, discrete circuits can readily be designed to accommodate voltage swings of ten or even fifteen volts—which is several times the highest input signal level the circuit will actually encounter.

## BUFFERS

Buffer amps are relatively easy to design, and when properly executed, contribute almost no distortion at all to the signal. Correct designs in this area should exhibit low or no gain, and should utilize discrete devices. High gain "op" amp ICs used as buffer amps suffer from the same liabilities as when employed as gain stages, namely limited bandwidth, lack of headroom, and high transient distortion.

## POWER AMPLIFICATION

Power amplification is the area where the four design philosophies assume really paramount importance, and in this section we'd like to discuss in detail how these philosophies are implemented within the actual circuit configurations.

Discrete Devices - Harman Kardon amplifiers are unique in their price range in having discrete circuitry at every stage of the analog signal path—in other words all of the gain stages of the amplifier through which the signal travels utilize discrete transistors exclusively to perform the amplification, for reasons discussed in the preceding chapter. True, many other manufacturers advertise the use of discrete output devices, but in general, where such claims are advanced, the first stage of the signal path will consist of an IC with all of its attendant problems.

## ULTRAWIDE BANDWIDTH

Ultrawide bandwidth properly refers to linear operation beyond the audio band, since there's little sense in increasing bandwidth if the amp can only generate high distortion out of band. Harman Kardon power amplifiers all exhibit full rated power response at very low distortion to beyond 100kHz. Such ultrawide bandwidth can generally only be gotten with low gain, low feedback circuits made of discrete devices. That's because high gain circuits with feedback loops are prone to instability if operated wideband, and in fact "op" amp that answers to this description frequently exhibits open loop rolloffs beginning as low as 100Hz. True feedback can bring the response back up again—up to a point—but that point rarely extends beyond 20kHz. Ultrawide bandwidth also requires selected output transistors, again a cost factor, and one most mass producers of audio equipment are unwilling to make.

## LOW NEGATIVE FEEDBACK

We have already examined rather extensively the generally adverse effects of high values of global feedback on sound quality. Now we would like to point out the design requirements in an amplifier using low values of negative feedback.

Low feedback amps must have low distortion before the application of feedback in order to perform acceptably. Modern performance standards call for distortion levels at the amplifier's output of at most a couple of tenths of a percent, and in a low feedback design this practically necessitates select high quality transistors to begin with, and tight regulation of the power supply in the voltage gain section to keep transistors in their most linear operating modes—all of which we do in our products. Furthermore, transistors must be biased so that they conduct some current at all times, which helps to eliminate switching noise caused by the transistor turning on and off in response to varying signal levels. All things being equal, the higher the bias level the more linear the transistor's operation, though linearity has to be balanced against heat build-up problems resulting from increased current flow. A low distortion low feedback amp is going to generate appreciable heat due to relatively high idling currents, and it must be built with adequate heat sinking and large capacity power supply to provide the idling current. It needn't run at the class A point where it's idling at full power, but it can't run ice cold either. Linearity in an amplifier generally bears an inverse relationship to efficiency which is why Harman Kardon designs for reasonably high bias currents in the output stages of our amplifier circuits.

We would add that in a low feedback amp considerable attention must be paid to the details of construction. Even passive components like solder joints and wiring can add distortion, and there's just no room for slipshod workmanship if you don't have a lot of feedback available for mopping up residual distortion.

## HIGH CURRENT

High current is the function of two aspects of amplifier construction, the number and ratings of the output devices, and the capacity of the power supply. Let's examine them both in turn.

The output devices, the power transistors that control the current flow and hence the power delivery of the amplifier, are strictly limited in the amount of electrical current they can deliver. If the current rating of a bipolar transistor—the type we use—is exceeded, the transistor is subject to failure. The internal resistance will drop to almost nothing as the transistor overheats, which in turn will permit more current flow, which in turn will generate more heat and so on until the transistor overheats and fails.

Any bipolar transistor will ultimately fail if forced to pass enough current, but commercially available units vary tremendously in their current ratings. Some power transistors are rated at under an ampere while others exceed twenty amperes, and naturally the big ones cost more money. Harman Kardon specifies heavy duty units, and we use plenty of them. In fact we build output sections rated for considerably higher currents than our power supplies can deliver. The output sections are massively overbuilt so they simply can't fail even in the case of an accidental short circuit.

Harman Kardon HK6950 R Integrated Amplifier

Building an amplifier with this kind of no nonsense output stage has another benefit which you should understand. We've already discussed how high current capability is necessary to exert adequate control over a loudspeaker. But the issue is really more complicated than that.

The output stage of an amplifier is where the rubber meets the road, the point where the circuit enters into contact with a relatively hostile electrical environment, namely the constantly changing impedance of the loudspeaker. Transistors are at their most linear when their outputs interface with a stable electrical load, and when the power supply from which they source current is electrically stable as well, and therefore the interface between the output transistors and the loudspeaker is always problematic.

When the loudspeaker is presenting the transistors with a low and/or reactive electrical load—a common occurrence during normal operation, the transistors will be taken out of their linear operating range, and will tend to generate more distortion. The feedback loop, the power supply, and in some cases, the bias circuit will try to compensate, but ultimately the design and capacity of the power supply will be critical in determining amplifier linearity in the face of difficult electrical loads.

The normal low or moderately priced integrated amplifier or receiver lacks the current capability to cope with such electrical loads because both the size of the power supply and the heat dissipation of the output devices are too low. In the presence of reactance in the speaker load, low current amplifiers may fail to maintain level frequency response, and they will clip at output levels far below their rated power. In fact the circuits of such amplifiers are perpetually poised on the edge of self destruction, and to protect them the manufacturers provide IV limiters—protection circuits which clamp the flow of current from the power supply when it exceeds a certain level. We hasten to point out that IV limiters are never used in Harman Kardon products.

We don't use IV limiters because they sound bad. Such limiters are constantly being triggered in the normal course of musical reproduction, and each time they clamp the power supply a burst of distortion results—distortion which does not show up on standard THD tests.

Harman Kardon amplifiers are provided with a very different kind of protection circuitry which is never engaged in normal operation, but only in the case of accidental short circuits. That's because it isn't needed otherwise since the output devices and power supply are equal to demands of real world loudspeakers.

## FURTHER WORDS ON POWER SUPPLIES AND HIGH CURRENT CAPABILITY

A power supply for a consumer audio component has two functions, it changes the alternating current from the wall socket into direct current, and it stores that current at a certain predetermined voltage to supply the transistors in the amplification chain. The energy storage capacity of the power supply is basically determined by the operating voltages and electrical capacitance values of the large filter capacitors which hold the electrical charge. Pricing goes up sharply with both voltage rating and capacitance, and this is often an area where manufacturers skimp, even the smallest Harman Kardon receivers have capacitor banks rated in the tens of thousands of microfarads.

Another area where our competitors tend to cut corners is in the power transformer which charges the capacitors. Transformers are rated in watts—conventionally expressed as volt/amperes—and again, the higher the rating, the higher the cost to the manufacturer. Our least costly receivers have power transformers exceeding 100 volt/amperes. Incidentally large transformers and filter capacitors tend to be heavy which further displeases the manufacturer of rack system components.

The most costly esoteric amps will have power transformers rated in excess of 1000 volt/amperes charging banks of capacitors with cumulative capacitance ratings in the hundreds of thousands of microfarads. We built just such a supply for our famous Citation XX, and there is no question that such supplies represent the preferred approach when final cost is no object. Such a brute force power supply can drive any speaker under any conditions with absolute stability and minimal distortion, and the amp which possesses such a supply will be able to double power delivery each time the load impedance halves.

## WHEN THIRTY WATTS IS BETTER THAN FIFTY WATTS

We position Harman Kardon amplifiers and receivers as the affordable high performance alternative to the typical mass market product, and we compete against the IC based mass market receivers rather than the cost-is-no object esoteric separates. Now many of these competing products offer what appears to be a better watts per dollar ratio than our own—in other words the customer seems to be getting more power for his money. thus to sell Harman Kardon successfully you must understand what standard FTC(Fair Trade Commission) power ratings really signify, and why high FTC power specs don't mean much in the real world.

The Fair Trade Commission established standards for rating power amplifiers back in 1975 due to the widespread publication of misleading or downright false power specifications by manufacturers. The FTC stipulated that the maximum power output of an amplifier must be maintained continuously into an 8 ohm resistive load over a period of several hours, and the manufacturer's own claimed maximum for total harmonic distortion must not be exceeded. The FTC also established a 4 ohm rating involving similar test procedures but utilizing a 4 ohm resistive load.

There's no question that the FTC standards brought an end to many of the more flagrant abuses in the industry and were ultimately beneficial for the industry, but in our opinion the standards didn't go nearly far enough, and that the test procedure itself does not emulate the actual demands of musical reproduction very accurately.

Here we'll review some of the concepts already discussed in the previous chapter on design philosophy, specifically those concepts supporting high current capability.

First of all neither an 8 ohm or a 4 ohm resistor is a very accurate model for the electrical impedance presented to an amplifier by most loudspeakers. A loudspeaker's measured impedance will generally include a strong component of electrical inductance at certain frequencies and the mechanical behavior of the loudspeaker in motion will simulate the effects of both capacitance and inductance. Inductance may be defined as the opposition to electron flow in a circuit, while capacitance is the opposition to a change in electrical charge. Either condition results in alteration in the phase relationship between current and voltage. In the case of inductance, voltage will increase while current lags, and in the case of capacitance, current changes as voltage droops. Since both voltage and current are required to generate electrical power, a purely reactive circuit with no element of resistance will not permit any electrical power to be developed. A loudspeaker is never a pure reactance, but at some frequencies reactance may be dominant, and in such cases the amplifier will have difficulty generating power.

As we indicated earlier, an amplifier's feedback loop responds to a rise or fall in output voltage by altering the voltage gain of the amplifier which in turn will tend to restore the voltage level to some nominal value at output. But maintaining output voltage in the face of a fluctuating impedance requires that the amplifier supply widely varying amounts of current at output. To maintain flat frequency response at high playback levels, an amplifier may have to provide up to 100 amperes of current with certain loudspeakers. This would represent an extreme, but 20 ampere current demands are not all unusual.

Now remember power, that is wattage, equals voltage times current. If an amplifier that is current limited is faced with a reactive speaker load, it may not be able to generate either high voltage or high current, and it's power output will be very low. To help you (Fig. 4a) understand this notion we've created a graph called a power cube which shows the actual power output of an amplifier under different conditions of reactance and resistance. The horizontal dimension of the cube shows plus and minus phase discrepancies in twenty degree increments. Zero degrees represents a purely resistive impedance while the positive direction shows increasing capacitance. The dimension of depth shows resistive

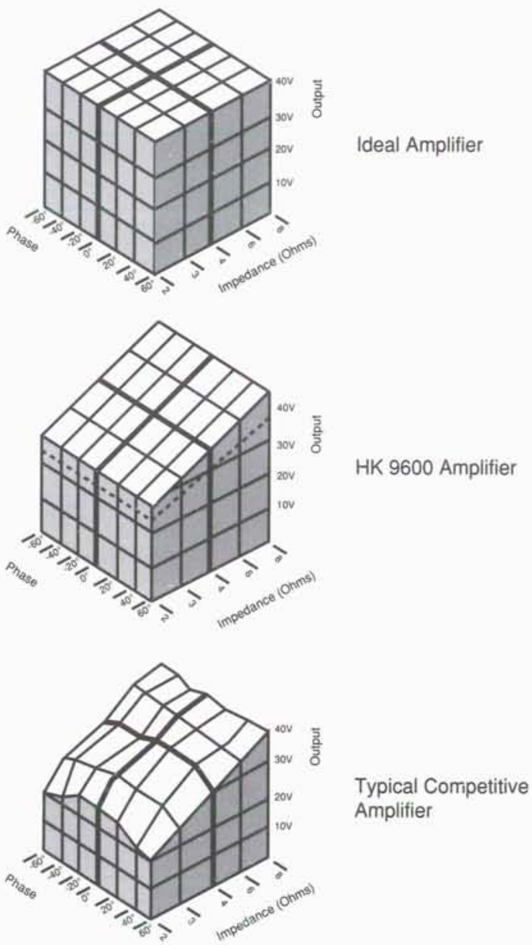


FIG. 4a

impedance, while the dimension of height shows the output voltage of the amplifier. You can see from the examples that competing mass market amplifiers will maintain voltage only into straight 8 ohm resistive loads. Since they can't dump current into the load either, then can't maintain rated power. They'll deliver the power into one ideal condition which will only prevail over narrow frequency ranges if at all in real world loudspeakers, and in all other conditions they'll fall short.

One way to explain the situation to your customers is to compare a low current IC amplifier with a high revving engine with no torque. Sure it can make high horse power with tall gearing, but it lacks the flexibility and a wide power curve. Similarly a low current amplifier with a high voltage rail on power supply will make power under under one set of constraints but cannot deliver power under varying conditions. It too can be said to have a narrow power curve. And that's why Harman Kardon amplifiers often deliver more useful power than competing products with twice the nominal rating.

So those are the basics, the principle reasons why our products are the best sounding reasonably priced amplifiers on the market. There is in fact a lot more involved in the designs we produce including some circuit refinements which are in the nature of trade secrets. We have in our employ some of the most experienced analog engineers in the world, and we gave them the mandate and the resources to strive for continual improvement.

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SECTION 5

# COMPACT DISC PLAYERS

In the compact disc player category, as in other product categories where it designs and manufactures products, Harman Kardon has concentrated on improving sound reproduction in the most cost effective manner. This chapter deals with the specific design strategies utilized to attain that goal with compact disc players.

The compact disc player is a very complex electronic device, and the details of its operation are far more involved than those for a tuner, cassette deck, or amplifier. That being the case, we will not concern ourselves here with a complete description of the CD signal path, but only with those aspects of digital reproduction where Harman Kardon engineers have been able to make improvements in sound quality.

**Determining Critical Differences in Compact Disc Player Performance**

If your professional experience with audio does not predate 1985 or thereabouts, you may not be aware that the compact disc format was the subject of considerable controversy among discriminating listeners at the time of its introduction. The extended dynamic range, low noise, and stable and widely separated stereo presentation of CD were readily appreciated by almost everyone who heard the new format. In spite of that, some listeners claimed that depth of image was poorly reproduced, and that overall sound quality was harsh, fatiguing, or unnatural.

Harman Kardon's engineering staff also heard problems with early players, and determined that with CD players as with so many other areas of consumer audio, published specifications were not telling the whole story. We decided to investigate the CD playback chain from beginning to end, locate the sources of the audible problems, and develop effective solutions within the limitations imposed by the format's design.

**How Can Digits Differ?**

Let us begin this discussion by stating that the compact disc format is an exceptional feat of engineering, and that the format's overall thoroughness is without parallel in consumer electronics. It brilliantly exploits the strengths of the digital encoding technique, and it permits an inexpensive, mass-produced device to achieve a level of pitch accuracy and freedom from noise and distortion that previously could only be approached by the most expensive professional mastering recorders.

It is worth noting that the conceptual work behind the CD format has as its goal the development of a format whose licensing requirements would set not only the basic, but also the ultimate level of performance, leading to almost absolute equivalence among players. Every player was *supposed* to sound the same and be essentially perfect.

Unfortunately, this did not happen, primarily as the result of an incomplete understanding by some designers of the nondigital aspects of the players including the transports, the D/A converters, and the analog output stages. When CD debuted in 1983, no one fully understood how to design the electromechanical and analog amplification subsystems within the players. Designers had only begun to explore the many possibilities in converter design. These were precisely the areas of focus for Harman Kardon's investigations.

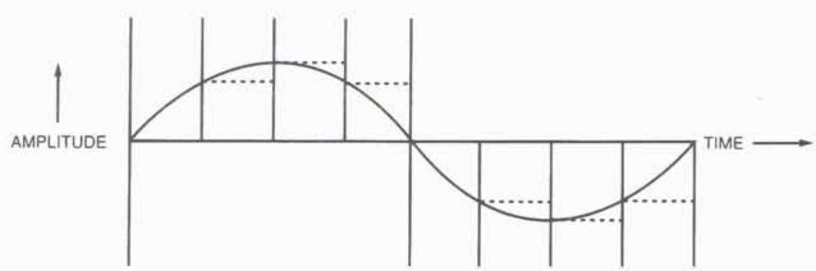
**Digital Fundamentals**

To fully comprehend our findings, you must have some basic understanding of how sound information is stored on the compact disc, and of the essential nature of digital recording.

When we say that the compact disc is a digital storage medium, what we mean is that digits or numbers are used to represent sound. In some sense a compact disc, or any digital recording, for that matter, is a sort of blueprint for the reconstruction and regeneration of soundwaves, rather than a direct record of the waveforms as in an analog recording (Fig. 5a).

The principles of digital recording are fairly simple, though the circuitry required to perform the recording operations is extremely complex.

The first step in constructing a digital record of sound is called *sampling*. In the process of sampling a sound, the sound is first picked up by a microphone which generates an analog electrical waveform as the sound causes the diaphragm to move. It is the electrical waveform, not the sound itself, which is actually sampled. That is done by delaying the signal in a capacitor at




The audio–signal amplitude is measured at each verticle line, and each measurement is converted into a binary word.

FIG. 5a

regular intervals, and taking a voltage reading of the capacitor's charge at each interval.

The timing of the intervals is critical because it determines the highest frequency the system can record. Since each half cycle of the wave must be sampled at least once, the sampling rate must be at least double the highest frequency recorded. In actual fact a slightly higher sampling rate is chosen because frequencies above the predetermined limit must be filtered out of the system. Also, no filter is perfect. The sampling rate for the compact disc format is 44.1kHz, and the nominal high frequency limit is exactly 20kHz.

The digital recording system itself may be considered to be a superfast digital volt meter with a memory. Each sample gets a voltage reading, and that reading is recorded in binary numbers the same way a computer stores information. The binary numbers form a code, and the digital recording technique itself is known as PCM (Pulse Code Modulation). The digital information may be stored magnetically on tape or a magnetic disc, just as a computer stores numbers, but in the compact disc format the storage medium is optical. The disc itself is etched with a microscopic spiral of pits whose varying lengths represent different numbers of pits. The pits themselves are cut to precise lengths with a recording laser onto a recording blank. The master is then used as a "mother disc" from which copies are made through a pressing process.

On playback, a tiny, highly focused laser beam is reflected off the surface of the disc into a lens assembly. There it is picked up by a photodiode detector which, in turn, converts the light energy into pulses of electricity. The pulses carry the digital code.

The code is massively redundant in terms of its information content so that data losses, unless very severe, shouldn't affect reproduction. The code also contains information for indicating the presence of dropouts, as well as data for reconstructing the signal accurately when dropouts do occur. Thus, by the time the information gets on the compact disc, the digital code containing it is far more than a simple numerical record of the samples. Several stages of decoder circuitry are required to pull the basic sample data out of encoding on playback.

Digitization itself is intended to make the signal virtually independent of the recording medium and largely independent of the accuracy of the electronic parts within the digital circuitry. Furthermore, the coding is designed to protect the signal from the effects of transport speed fluctuations and microphonics.

To a considerable extent, digitization does all of these things.

Thus, the purely digital signal circuitry within a compact disc player tends not to vary much from player to player. As a result, it has not been the focus of designers attempting to improve the sound reproduction of the CD players, with the exception of the error concealment circuits, which tend to be more powerful in expensive models. It is at the beginning and the end of the signal chain—at the digital pickup and the digital-to-analog converter—that players began to diverge from one another in performance. These areas are where Harman Kardon has chosen to concentrate its design efforts.

**Transport Mechanics**

All Harman Kardon compact disc players use highly accurate motors for rotating the disc and advancing the laser pickup across the disc surface. Typically, the motors in our players are much more precise than those used by our competitors in similarly priced players. Transport accuracy has a direct bearing upon the number of uncorrectable errors that will occur when the pickup is reading the disc. Of course, uncorrectable errors will result in inaccurate sound reproduction.

In addition to employing premium quality motors in our CD mechanical systems, we further reduce error generation at the pickup by shock mounting the transport. This reduces chassis vibration which could disturb the transport system during operation.

**The Digital-Analog Interface**

A compact disc player is a digital audio device only through part of its own internal signal chain. It picks up the audio information from the disc in digital form, and it performs several decoding operations within the digital domain. At one critical point in the circuit the digital code is converted to an ordinary 20Hz – 20kHz analog form, and all subsequent treatment of the signal is strictly analog.

The device that converts the digital information to an analog signal is called a D/A (digital-to-analog) converter. It probably represents the most critical single stage of the signal path, as the design and quality of the converter has a major effect on sound quality. The line amplifier stage that follows the D/A is also fairly critical. Unfortunately, it represents an area where other brands are likely to cut corners.

Obviously, a brand that is concerned about sound quality will use the best available D/A converters and line amplifier for the price, but the quality of these individual sections is not the sole criterion that determines the overall sound quality of the deck. Equally important is the way that the total circuit is laid out on the circuit board, and how the power supply is constructed.

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Together these three design aspects, D/A converter design, line amplifier, and board layout and grounding, all help determine a CD player's performance.

**Electrical Isolation, a Key Principle in Designing for Sound Quality**

The digital circuits used for processing data consist of complex arrangements of electrical switches. The switches themselves are either open or closed, and the circuits they control are either "open" and conducting current or "closed" and shut off. The act of opening or closing the switch creates bursts of electrical noise with a concentration of energy in the region above 20kHz. That noise can seriously degrade sound quality if it is allowed to leak into analog audio circuits. To get an idea of what this noise can sound like, place a portable AM radio close to a computer. The noise or "hash" you hear is the same thing that can pollute the quality of audio in a CD player.

Harman Kardon FL8400

If you haven't had a lot of exposure to digital audio theory, you might be tempted to ask why the switching noise doesn't contaminate the digital signal itself. The answer is simple. A digital audio signal is essentially a code of instructions to the digital to analog converter. The instructions themselves are transmitted in an on/off binary code. As long as "on" is clearly distinguishable from "off," the presence of electrical noise in the digital circuit is irrelevant. In an analog circuit, on the other hand, the presence of electrical noise has an immediate effect on the sound quality. That's because in analog the music signal is transmitted in a continuous waveform whose precise shape must be preserved throughout the signal path. Switching noise alters the shape of the waveform, and thus affects the music signal.

A digital circuit inevitably creates lots of electrical noise while performing the functions for which it was designed. It is noisy by its very nature. For that reason, the digital circuits in a CD player must be isolated from the analog output circuits as much as possible.

Unfortunately, isolating one from the other isn't as easy as it sounds. The high frequency alternating currents that comprise digital switching noise often find paths through the chassis, ground circuits, and power supplies—anywhere a voltage potential exists on the component.

Harman Kardon takes extraordinary measures to prevent this switching noise from reaching the analog circuitry. We do that by dealing with the noise at both of its sources, the digital audio circuitry and the digital display on the front panel. Both the display and the digital audio section have their own regulated power supplies, their own grounds, and their own secondary windings on the power transformer. We even use shielded cables for electrical connections within these separate sections so electrical noise can't be radiated or induced. No one else is taking these kinds of precautions in comparably priced CD players.

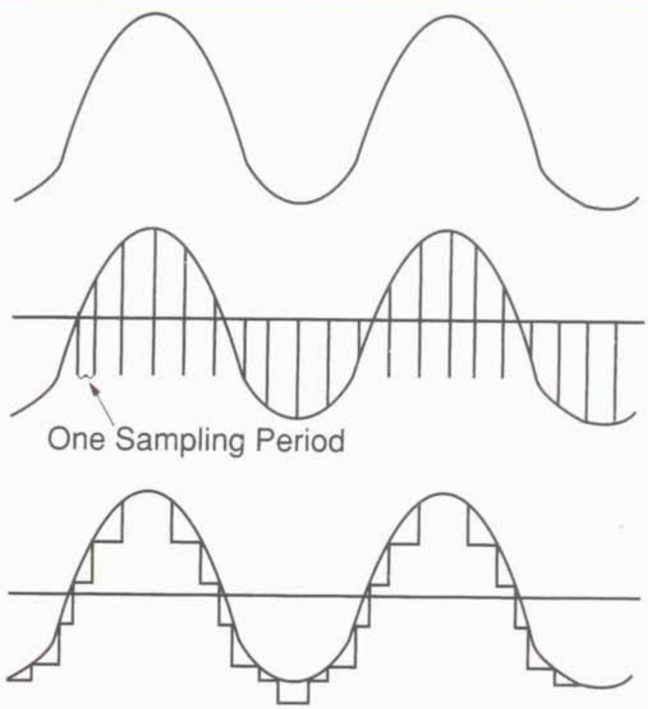
In addition, we provide separate transformer windings, regulators, and grounds for the transport motors because motors also generate electrical noise and we don't want our customers to hear that noise either.

## THE D/A CONVERTER, THE VITAL LINK

We have already stated that the D/A converter is the most critical element in a compact disc player. It is also the weak link in the signal chain, the part of the circuit which is most in need of further development and refinement. All of D/A converters in use today are imperfect devices, but some perform their function much more successfully than others.

### Old Technology - The Conventional Switched Resistor D/A Converter

The most common form of D/A converter in use in compact disc players is the *switched resistor* or ladder type of which there are several variants. This device performs the conversion by taking the sixteen bits of information forming the digital code for signal intensity, and using each bit to control its own separate resistor, which in turn is placed in series with a constant current source. Each of these switched resistors passes a different value of electrical current, and each value corresponds to the magnitude represented by that particular bit within the digital code. The currents released when all the resistors are switched are summed, and that sum appears as a series of individual flat topped pulses—actually square waves. Sequences of these pulses look like an up and down staircase (Fig. 5b) and follow




Staircase waveforms.

FIG. 5b

the general contours of analog waveforms. At the output of the converter an analog filter sheers off the high harmonics which form the sharp corners in the "staircase" form. This results in rounded waveforms that approximate the original analog signal.

In most current CD players using switched resistor D/A converters, the sixteen bit information is sampled repeatedly, that is, it is "oversampled" four or eight times so that the sequence of pulses is made very dense, and the width of the steps in the staircase is reduced. The effect of this oversampling is to move the harmonics representing switching noise to a frequency range which is several octaves above the range of human hearing. That permits the manufacturer to use a relatively simple analog filter in the audio band, which in turn tends to minimize audible phase shift.

Switched resistor D/A converters can work extremely well, but there's also a lot that can go wrong with them. The accuracy of decoding is dependent upon the accuracy of the individual resistors, and for best results each converter should be hand calibrated. Naturally, very few customers are willing to pay for converters where each individual bit resistor has been hand-trimmed.

There are other problems. Usually an additional stage known as an *aperture circuit* or *output sample and hold* must be included between the D/A converter and the analog output filter. Aperture circuits serve a purpose, but they also complicate the signal path. When combined with a complex output filter, the analog signal path becomes very convoluted. Suddenly claims of transparent digital storage no longer seem so convincing.

## BETTER D/A CONVERSION

The performance of conventional switched resistor D/A converters appear to be very fundamental, and they are not subject to radical improvements. True, there are "high bit" schemes where the converter itself is endowed with resolution beyond the sixteen bit requirement of the format. These may allow for a higher degree of decoding accuracy within the basic switched resistor configuration, but the critical high frequency noise and filtering problems remain. Oversampling, which has increased from the mere doubling used in second generation decks to an eightfold multiplication in the fastest current generation ladder converters, can move the switching noise up the audio spectrum, but it can't decrease the quantity of noise at all—in fact the opposite will occur. Thus standard design tactics for varying ladder decoder parameters can only provide best fit compromises, they can't attack the performance problems at their root.

Two types of converters have been developed which appear to offer some ways out of the design impasse confronting the digital audio engineer. The first of these, known generically as single bit, has been fairly widely adopted by the industry, though Harman Kardon was the first manufacturer to offer the technology across the product second technology, called RLS™ (Real Time Linear Smoothing), was developed by Harman Kardon and is protected by international patents.

**3D Bitstream™ D/A Converters, Precise and Consistent**

3D Bitstream D/A Converter (Fig. 5c), a type of single bit design exclusive to Harman Kardon, combine an extreme degree of design elegance and simplicity with numerous performance benefits. They entirely eliminate the network of switched resistors characterizing the ladder type converter, and remove the requirement for the output sample and hold circuit. Such converters

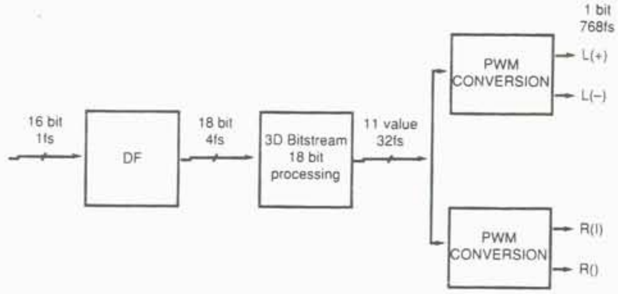



FIG. 5c
Block diagram of 3D Bitstream.

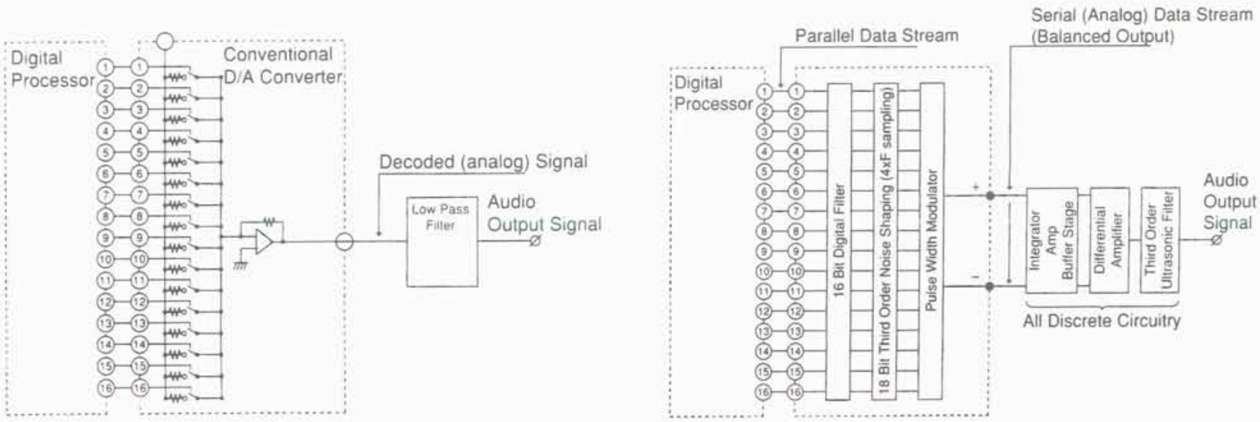


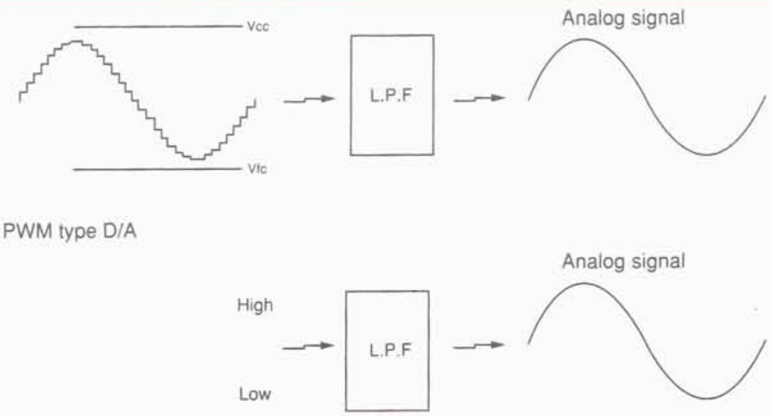

FIG. 5d

output filter to be used in place of the complex multipole filters which appear in most ladder designs. Also, the precision of a bitstream converter is inherent in its design, and does not depend on meticulous tweaking and individual adjustments. Finally, bitstream conversion pushes the high frequency switching noise much higher up in the spectrum than has previously been possible, to a range where filtering requirements become far less stringent. It does that by what is essentially massive oversampling.

The single bit process avoids converter design complexity while achieving the equivalent of oversampling rate in the hundreds, by first translating the PCM digital audio signal into another type of digital modulation scheme known as PWM (Pulse Width Modulation).

Pulse Width Modulation is a modulation technique where the audio signal is imposed on a carrier frequency located (ideally) several octaves above the audio band. The carrier appears as a series of flat-topped pulses whose widths are made to vary by the modulation of the audio signal (Fig. 5d). Pulse width modulation is closely related to frequency modulation since information in the modulated signal is carried in the form of *variations in the duty cycle of the carrier*. The amplitude of the signal can only vary between a maximum and minimum value with no intermediate values. The pattern of variations in the width of the pulses represent the smooth waveforms of the audio signal. It is almost as if the analog waveform has been rotated and expressed within a third dimension.

Pulse width modulation itself may be either a digital or an analog technique since the width of the pulses may be infinitely variable or variable according to a series of fixed intervals. When a PCM




Resistor ladder type of D/A conversion versus
PWM type of D/A conversion
FIG. 5e

encoded signal is used as the basis of a PWM encoded message, the width of each pulse must vary according to fixed intervals which themselves are determined by the PCM code for each sample of the signal. In other words, each binary number on the compact disc (Fig. 5e) signifying signal intensity for a given sample, can also be used to assign a specific *width* to a PWM pulse. The details of how this is done could fill another chapter, but essentially the simultaneous sixteen bit block of code for each sample is broken down into a new single bit code which is transmitted very rapidly, one bit at a time. This *bitstream* provides the information for assigning the duty cycle length of each PWM pulse. So in this case PWM becomes a digital technique, and the conversion process involves a digital to digital conversion *before* the conversion from digital to analog.

What are the advantages of making the translation to PWM before going to straight analog? In fact, there are several good reasons for adopting this approach. The first is that the output of the converter is defined at the one bit level. Instead of a staircase with steps of different heights, you get a stream of pulses of varying widths but of exactly the same height. Only one digital bit of information is needed to define a single height level, hence the term "single bit system". Since one, and only one, bit is needed to represent the data, a ladder of ultra-precise switched resistors representing different bit values is no longer needed to convert the data into an analog waveform. In fact, all one must do is run the bitstream through a fairly simple low pass filter and an analog waveform will come out, as most of the conversion process has already been accomplished within the digital domain.

There is one small catch to all of this. The pulse train in a PWM scheme must be very fast because the audio signal itself must be massively oversampled. (The individual samples must be re-done several times.) This is because the PWM width variations are derived from the original PCM signal by creating long strings of one bit pulses whose *averaged values* have a similar range of gradations as the simultaneous sixteen bit samples of PCM. In other words, the instantaneous intensity information stored in the PCM code is spread out in time, and in order for the process of sample conversion to happen essentially in real time, the converter circuitry must be extremely fast.

Circuitry with the requisite speed simply wasn't available in off-the-shelf microchips until rather recently, and so the bitstream technique wasn't feasible in first generation players. Now it has become a much more practical technology for achieving high accuracy decoding than any type of conventional switched resistor ladder circuit. Designing a highly accurate digital clock circuit to provide the massive oversampling of bitstream is much easier than engineering a precise switched resistor ladder circuit. Reasonably priced bitstream decoders easily rival the performance of the most expensive ladder types, and they provide just the sort of marriage of high performance with high value which is a requirement for all Harman Kardon products.

Specifically, the bitstream converter achieves low level linearity and phase response that is as good or better than that of the most exotic ladder converter. In short, bitstream is a very logical choice in today's digital audio technology.

As you'd expect, MASH-bitstream performs optimally when used in a player with excellent mechanical and analog electronic subsystems. Here again, Harman Kardon has achieved a unique synergy. We are the first and only manufacturer to use what we call 3D bitstream, which means that bitstream conversion is combined with discrete output circuitry (the "3D" stands for "*D*igital to *D*igital," referring to the digital conversion which takes place within the decoder, and *D*iscrete circuitry).

Nevertheless, for all its design elegance and performance advantages, 3D Bitstream has been surpassed by a more radical conversion technology. One which, as it happens, uses precisely the type of high bit ladder configuration which our 3D bitstream has been steadily edging out of the market. That technology is called Real Time Linear Smoothing, or RLS.

**RLS Challenges the New Conventional Wisdom**

Make no mistake, we at Harman Kardon are not ready to retire 3D bitstream. 3D bitstream has set new standards for musicality while proving highly cost-effective. Though the performance of bitstream is surpassed by the newer technology of RLS, it is still offers very good performance for the price. RLS decoders will remain more expensive to build than the bitstream type, and thus they will be limited to use in high end models.

Nevertheless, RLS is absolutely better than bitstream, and Harman Kardon's premium players with RLS offer the best decoding performance available at any price.

**Up the Ladder**

Before we discuss the actual operation or RLS, consider for a moment the limitations of the single bit approach, and why we were driven to develop our new RLS technology.

Very generally, single bit conversion offers *conversion accuracy* on a par with that of the best conventional ladder designs, and it does so much more cost-effectively. It also pushes switching noise much further away from the audio spectrum. Unfortunately, it actually increases the total value of the noise. That's because the PWM pulse train resulting from the conversion takes the form of a series of high amplitude spikes with relatively huge amounts of high frequency energy instead of a staircase. Granted, we have found that PWM switching noise is easier to filter than the lower frequency noise components created by the staircase, but still a single bit system cannot be considered optimum in regard to the noise levels it produces.

Now let's look at the very different kind of noise suppression strategy used in RLS.

Our RLS decoding circuit begins with four very high quality 18 bit, 8 times oversampling converter chips—two per channel. The chip itself is used in some other premium players, though normally only in pairs—one per channel.

The 18 bit, 8 times oversampling converter represents the current state of the art in ladder architectures. Compared to older ladder converters—though not to the MASH-bitstream type—it offers very superior low level linearity, phase response, and noise suppression. Since this type of "high bit" converter has gotten a lot of publicity in the industry, we think you should know how the conventional units perform before we discuss the improvements incorporated in our RLS "hotrodded" version.

An 18 bit linear converter works as follows:

The high bit converter takes the 16 bit data on a disc and passes it through an ultra high speed switched resistor ladder circuit having 18 bit precision. The converter multiplies both the number of samples and the number of bits. This has the effect of narrowing both the width and the height of the individual "stairs" in the staircase, thus increasing their number. More stairs means a higher switching rate, and thus a higher


Harman Kardon HD 7725

frequency range for the noise components.

Using 18 bits to decode 16 bit data has another benefit as well, since the full resolution of the converter need never be used. Normally the first and sixteenth bits, the MSB (most significant bit) and the LSB (least significant bit), are the most subject to conversion inaccuracies. With an 18 bit converter, it is not necessary to use either the MSB or the LSB when converting 16 bit data. The 18 bit ladder circuit cannot actually increase the resolution of the disc itself—it's still 16 bits and no more—but a linear 18 bit ladder can offer greater accuracy in sixteen bit decoding than a 16 bit circuit, simply because it's never pushed to its limits.

In and of themselves, 18 bit decoders still suffer from high frequency switching noise, and the extra increment of resolution does nothing to remove such noise. Eighteen bit resolution combined with 8 times oversampling moves that noise further into the ultrasonic region, but it does not reduce its absolute value. Indeed, bitstream techniques move the noise components quite a bit further up in frequency, and thus hold a clear advantage in that regard. RLS technology, which is possible only with ladder decoder architectures, throws the performance advantage decisively back into the ladder camp, because RLS — and RLS alone — greatly reduces the absolute value of the switching noise instead of merely shifting its frequency range.

RLS reduces switching noise by altering the shape of the waveform at the output of the D/A converter *without electrical filtering* (Fig. 5f). All ladder type D/A converters manage to convert the initial staircase pulse train emanating from switched resistor ladder into smooth analog waveforms, but most do so by means of electrical filters. These

## Real-time Linear Smoothing

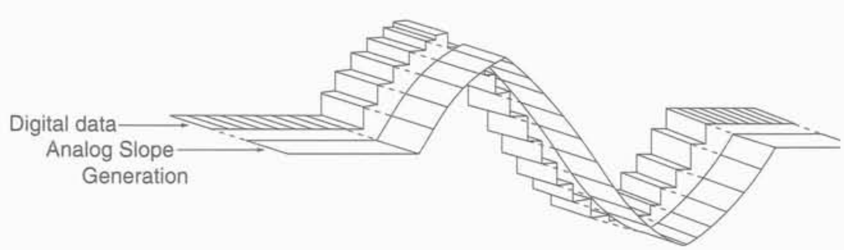



FIG. 5f

filters do not eliminate noise, they merely shunt it to ground. Since other circuits share the same ground plane, they are susceptible to picking up such switching noise—though admittedly good circuit layout can do much to minimize the damage. On the other hand, RLS eliminates the problem altogether, and to that extent it is a real breakthrough.

So how does RLS smooth the staircase and banish the noise at the same time?

Basically a seven step process is involved.

Step 1: We start by taking the information in either of the stereo channels, duplicating that information, and sending a copy to each of the two 18 bit, 8 times oversampling converters assigned to that channel. (Since the information in each channel is processed in exactly the same way, we'll confine the rest of the discussion to just one channel with its two separate converters.) Why two converters per channel? A look at the two succeeding steps in conversion operation will provide an explanation.

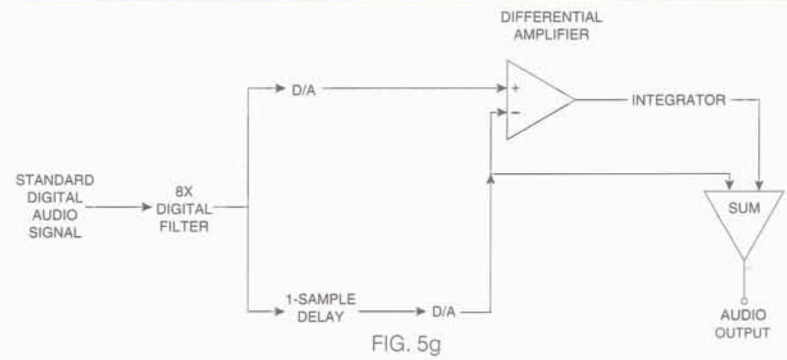


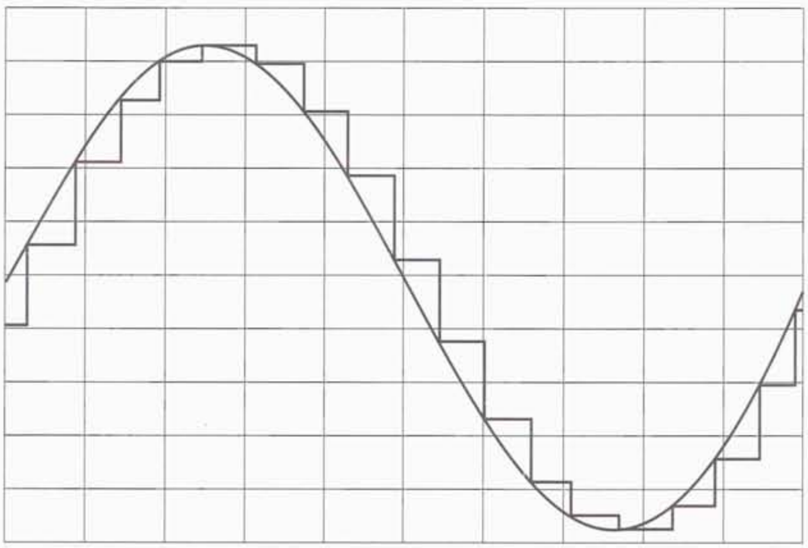
FIG. 5g

Step 2: After the digital audio signal has been duplicated and the duplicates sent on their way, one of the duplicates is delayed one half sample in relationship to the other so that the input of one converter is exactly one half sample behind that of the other. The outputs of the two converters are now exactly one sample apart as well.

Step 3: Both converters operate in the usual way on the duplicated data, and they produce the usual staircase output signal. Of course, the two outputs are out of sync by one sample length.

Step 4: The staircase output of the lagging converter is then split again, and one half of that signal goes to a *comparator-integrator* circuit, while the other half goes to a summing amplifier or mixer. Meanwhile the staircase output of the leading converter, the one whose input hasn't been delayed, also goes to the comparator-integrator circuit block (Fig. 5g).

Step 5: The comparator portion of the comparator-integrator is what is known as a differential amplifier, that is, its output can only be the difference between two input signals. In this case those inputs are the outputs of the two converters, two pulses representing two successive samples. Thus, the output of the comparator is the



A 20kHZ , 0dB signal re-constructed by an 18 bit D/A converter with an 8x digital filter, before and after RLS filtering

FIG. 5h

difference in voltage potential between the two samples.

Step 6: This *difference signal,* which itself takes the form of a square wave, is then sent to the integrator portion of the integrator-comparator circuit block where it is converted into a triangle wave (integrator circuits are commonly used as triangle wave generators in test instruments). Here's where the magic of RLS appears because that resulting triangle wave, *when added back to the lagging pulse,* will completely and perfectly fill the triangular notch separating the lagging and leading pulses! Do this again and again, and thereby fill all the notches between stair-pulses, and the stairway itself disappears (Fig. 5h).

So how do you add the triangle wave to the square wave representing the lagging sample? That's where the summing amplifier comes in during the seventh and last step of the process.

Step 7: The summing amplifier is a relatively simple device designed to add two wave forms together. It puts the triangle wave right on top of the lagging pulse, and the process is complete.

By producing a signal representing the difference between two samples, and placing that difference signal between two successive samples, the converter is *interpolating* data, or filling in missing information which would have been supplied by a wideband analog recording device. The converter is also eliminating high frequency noise because that noise is represented by the sharp edges defining the individual pulses. When the spaces between the pulses are filled, those edges are eliminated and, with them, most of the noise. It's almost like adding numbers to a paint by numbers sketch. The outline becomes more natural, detailed, and realistic.

RLS converter are not perfect devices, and they cannot eliminate all digital noise from the playback system. However, they do substantially reduce it, so much so that a simple first order filter consisting of a single capacitor is all that is required at the output of the converter section. By simplifying the filter, we are eliminating the single biggest source of image degradation and blurring of instrumental timbres. Basically, RLS is a clever fix for a standard conversion scheme rather than a radically new conversion technology as was bitstream.

The important thing, however, is that it works, and it works marvelously. Since we have developed RLS and perceived its benefits in exhaustive listening tests, we have become convinced that high frequency switching noise probably accounts for most of the criticisms of digital sound quality that have been persistently voiced by audiophiles since the CD format was introduced. By practically eliminating that noise, we have succeeded in refining digital sound reproduction to the point where its full potential is finally realized.

## THE OUTPUT AMPLIFIER

The line level output amplifier in a compact disc player has at least as much impact on sound quality as the output amplifiers in cassette decks and tuners, and perhaps more. As you now know, a compact disc player may generate high frequency switching noise. This creates intermodulation products in a poorly designed line amplifier. The use of low bandwidth chips at output stages, combined with poor digital isolation, is one of the primary factors in the harsh sound quality associated with budget CD players. From our first entry into the CD player market in 1985, Harman Kardon has never made a player which did not include a fully discrete output section, and that's true of our CD changers, as well. Discrete circuitry forms the third D in the 3D bitstream nomenclature, and it is a prime factor in the superior sound quality of our players.

## SYNERGY IN DESIGN

Our long term research into CD player performance has indicated that significant improvements in sound can not be had by exclusive concentration on one aspect of design. 3D bitstream decoding and RLS are seminal improvements because they eliminate several sources of potential inaccuracies. They result in a notably cleaner signal at low levels where problems in a digital audio system are most likely to appear.

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Isolating power supplies, which we also do, has to be considered a complementary design strategy, because it too reduces the magnitude of low level artifacts. The same might be said of our insistence upon using high speed, low feedback discrete components in our analog output stages.

In summary, Harman Kardon design techniques result in CD players that are free from harshness and graininess in reproduction. They reproduce voices and instrumental sounds more convincingly. Ultimately, then, they deliver exactly what everyone wants from high fidelity equipment.

6

SECTION 6

# CASSETTE DECKS

6

In this chapter we'll discuss those aspects of cassette recorder design which make for superior performance, and examine in detail how Harman Kardon cassette decks achieve their exceptional performance. Specifically we'll go over the principles of magnetic recording, tape equalization, tape bias, and the Dolby noise reduction systems.

## THE NATURE OF THE DEVICE

The cassette deck differs from the other component categories we've considered in that it is a magnetic and an electro-mechanical system as well as being electronic. Unlike a CD player, which is electromechanical also, though not magnetic, a cassette deck is entirely analog, and as such it is incapable of error correction or error concealment—the sort of computational strategy for dealing with losses in the recording medium which are feasible in the digital domain. In a cassette tape deck—or in any other analog playback format for that matter—the accuracy of reproduction is wholly a function of the accuracy of the transducer and the transport system. Thus there is no way to make a superior cassette deck cheaply. The mechanism must be precisely machined and aligned, and superior structural materials must be utilized. Construction quality is everything, and there is no room for manufacturing shortcuts.

Harman Kardon happens to excel at high tolerance manufacturing which has given us a decided competitive advantage in the assembly of premium quality cassette decks.

## CASSETTE IN CONTEXT

Before we discuss how we build cassette decks and assess their performance, we'd like to make a few observations on the current situation and probable future of the cassette medium.

Cassette began in the early sixties as a format designed for office dictation. It's still used for that purpose, and it probably will continue to be so used for many years to come. Cassette began to emerge as a music format as far back as the mid sixties, but it didn't gain immediate acceptance because of the rather poor fidelity of the earliest units. Then in 1971 Harman Kardon demonstrated that an optimally designed and constructed cassette deck with Dolby noise reduction could approach the sound quality of the then dominant medium, the stereo phonograph record, and the music industry took notice. Harman Kardon was the first manufacturer to utilize Dolby B noise reduction, and due to our efforts and those of several other manufacturers of high quality cassette recorders, the medium quickly established itself in consumer high fidelity recording applications. In fact, by 1980, it had almost wholly succeeded in displacing the open reel tape deck among home recordists.

For ten years the cassette was the only real recording medium available to the consumer in the U.S. and was the dominant playback medium in automotive and personal stereo. And for many years cassette not CD, has enjoyed overwhelming dominance in sales of pre-recorded software. Nevertheless, the cassette format is facing strong challenges at the present time from at least two emerging formats, and some in the industry have expressed doubts as to the cassette's continuing viability.

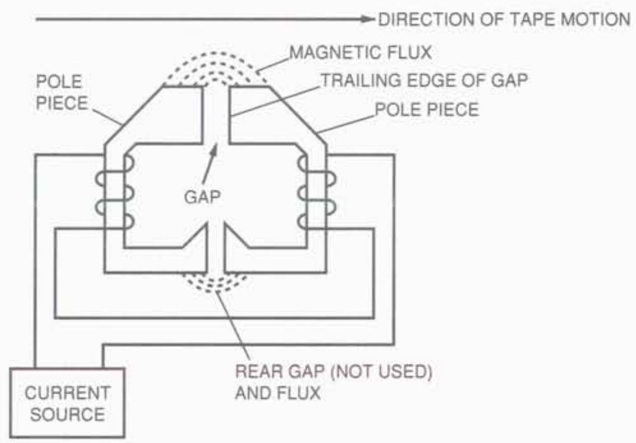
In an attempt to cast some light on this matter, we will say just a few words concerning these challengers and possible successors to the cassette.

In 1990, after many delays, DAT (digital audio tape), a cassette like digital tape format announced back in 1987, was finally introduced into the United States as a legitimate consumer format after earlier winning some acceptance in semiprofessional applications. We won't go into a discussion of DAT operating principles. Suffice it to say that DAT works, but, on the other hand, for various reasons having nothing to do with technology per se DAT has failed to make much of an impact in the consumer marketplace thusfar, and prerecorded software remains critically limited. Subsequently other manufacturers announced new competing recordable digital formats for consumers, one utilizing a small optical disc, and another a cassette sized magnetic tape cartridge. At the time of this writing, neither format has actually been released, so for all the talk of cassette obsolescence, nothing has been established to take its place. Thus we at Harman Kardon see a market for high quality cassette decks continuing for some years to come, particularly inasmuch as Dolby has recently introduced a new and very much improved form of noise reduction called Dolby S that enables a cassette deck to rival digital dynamic range.

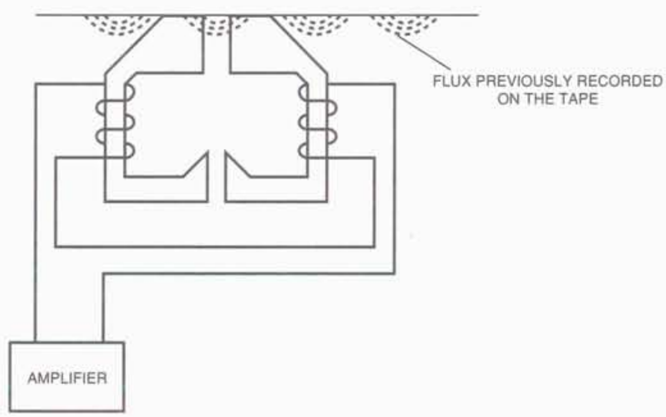
## THE NATURE OF THE MEDIUM

A cassette deck is essentially a miniaturized version of the reel to reel tape decks used in recording studios. The principle of operation is exactly the same: an audio signal passing through an electromagnet varies the magnetic flux between the magnet's pole pieces which imparts varying degrees of magnetism in the thin layer of powdered metal or metal oxide on the tape passing beneath the magnet. Playback is exactly the reverse. The magnetized tape itself generates magnetic flux at all times, and as the flux lines intersect the magnetic circuit of the playback head, a flow of electrical current results. The actual generation of a signal during playback occurs as a result of changes in the average level of flux in the magnetic gap of the playback head.

Both the playback head and the record head of a magnetic tape recorder are roughly similar in form. Both consist of two C (Fig. 6a) shaped magnetic cores separated by narrow magnetic gaps and wrapped in a coil of conductive wire which carries the audio signal. In the record head the conductive wire generates magnetic flux which circulates




A Record Head.




A Playback Head.

FIG. 6b

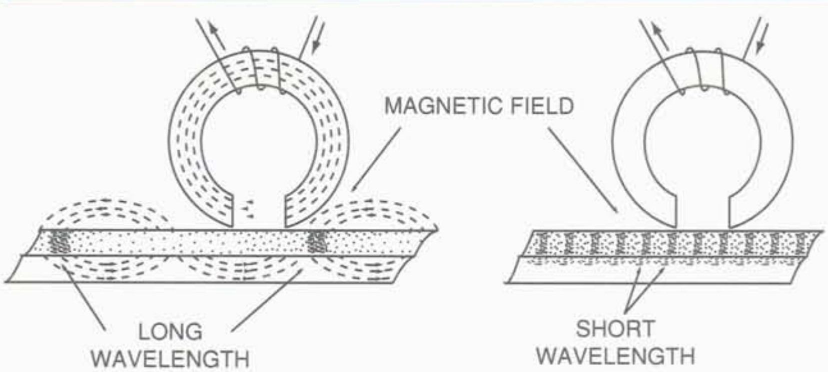



FIG. 6a

through the cores as is imparted to the recording tape through the gap. In the playback head, the magnetic fields in the tape itself cause a flow of electricity through the coil as the lines of (Fig. 6b) magnetic force intersect the coil—in other words, the magnetized tape and the playback head coil function as an electrical generator.

But while a magnetic tape recorder is extremely simple in principle, achieving good fidelity from a magnetic recorder demands very sophisticated design work and ultraprecise manufacturing. That's because the basic medium of magnetic tape is highly nonlinear, and because the magnetic field itself is difficult to focus, while the moving tape is hard to stabilize in relation to that field. The extent of the difficulties can be grasped from the fact that more than forty years elapsed between the time the magnetic recorder was invented in 1898 and the appearance of the first quality recordings during World War II.

Unfortunately the design of the perfected reel to reel tape recorder provided an inadequate model for designers of high fidelity cassette decks. The professional recording deck was designed to achieve the very highest level of fidelity irrespective of size, operational convenience, and tape expenditure. The initial cassette format, on the other hand, was intended for nothing more than telephone sound quality with operating convenience and economy as its main selling points.

Now let's examine for a moment the specifics of how the two formats differ and the consequences those differences have for performance.

On a microscopic level "analog" tape is really a digital medium. Within the magnetic coating on the tape surface are a multitude of tiny particles, each of which is itself a magnet with its (Fig. 6c) own north and south poles. Each of these tiny magnets is known as a domain.

Now when the tape is magnetized, the poles of these little domains are oriented within the magnetic field produced by the recording head. On the positive half of the wave cycle the magnets line up south/north in the direction of tape travel, and on the negative half cycle they take on a north/south orientation. The number of polarized domains in a given segment of tape ultimately determines the strength of the magnetic flux field on the tape, and when all of the little magnets are

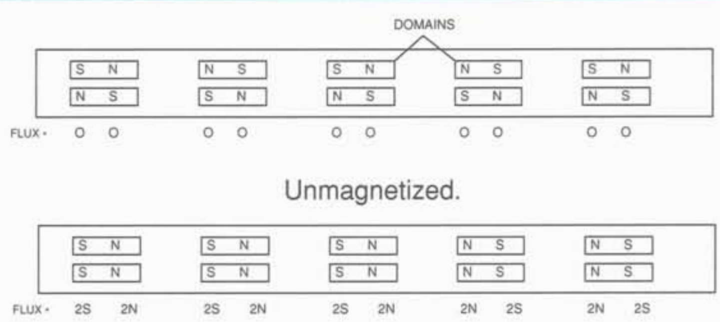



FIG. 6c

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aligned in the magnetic field, the surface layer is fully magnetized and at that point increasing the field strength from the recording head will not increase the flux density on the recording tape.

When this condition of maximum flux density occurs, the tape is said to be saturated, and such a state of saturation marks the upper limit for the signal intensity that can be captured on the tape. From that point the only way to increase flux density is to cause the tape to pass by the recording head more rapidly in other words, to increase tape speed, or to increase the width of the tape, or both. Either or both strategies will bring more domains under the record head within a given time span, which in turn will increase the flux density in the gap.

In this light a comparison between cassette and the reel-to-reel tape format in terms of tape speed and tape width is very instrucive. Professional two track open reel tape recorders use tape of up to one half inch in width traveling at speeds as high as 30 inches per second. Cassette tape in contrast is a little over one eighth of an inch in width and travels at a snail's pace—one and seven eighths inches per second. There's simply no comparison in information density between the two formats—cassette isn't even close—and thus the signal level cassette can accept without saturating is very much lower.

But slow speed and restricted ability to store information are far from the cassette's only limitations. The basic form of the cassette itself is the source of many problems. Whereas open reel tape can be transferred from reel to reel, and the reels themselves can be made very precise even to being spin balanced, the cassette housing is integral to the format and represents a given, and generally a liability in terms of performance. The cassette housing carries the tape on loose fitting internal spindles within a thin gauge plastic housing. Worse, there's usually considerable play in the cassette's moving parts, and an imprecise tape path through the housing. And because of this imprecision in the tape holder, it is difficult to get the tape to move smoothly past the magnetic heads.

So in total the cassette medium is compromised by very slow tape speed, narrow track width, and a less than optimal mechanical system. Even so, a Harman Kardon cassette deck, when equipped with Dolby S, and when utilizing high quality blank tape, can produce a copy of a studio master which is difficult to distinguish from the original. That's because none of the individual liabilities is insurmountable, and indeed Harman Kardon's Dolby S decks exemplify just what is possible with cassette when all operating parameters have been optimized.

## TAPE SPEED AND HIGH FREQUENCY RESPONSE

Let's take a closer look at the most fundamental of the cassette format's liabilities, the information density problem arising from slow tape speed and narrow track width. This basic limitation affects both the frequency response and the dynamic range of the cassette medium. Depending on tape formulation, a premium cassette deck without noise reduction will have a best case signal to noise ratio of less than 50dB—certainly nothing to write home about. Frequency response is much harder to quantify because it is subject to a multitude of variables, but we can say with some confidence that it is very, very difficult to get 20Hz - 20kHz response at the -10dB recording level specified for professional reel to reel decks. At the same time, you should note that while cassette may be said to be digital on the microscopic level, it is in no sense a digital modulation recording system and thus there are no hard limits on frequency response. It is possible though not easy to extend the high frequency response of a cassette to beyond 25kHz. In other words, there are no brick wall limitations—merely a point of diminishing returns.

On the low end, the cassette medium is limited in frequency response by the magnetic saturation characteristics of the recording head. Like the tape itself the head can only support a certain magnetic flux density and when the electrical energy generating the field increases further, the field does not change in response. Thus bass response tends to drop off gradually at low frequencies in the cassette medium, with the roll-off generally beginning somewhere around 40Hz.

In the high frequencies, playback response is strictly a function of the width of the magnetic gap in the playback head and the alignment of the gap in relation to the tape itself. Unless the gap is narrower than the magnetized portion of the tape passing beneath it—in other words narrower than the segment of intense magnetization representing the wave peak, then that magnetized segment simply won't generate an appreciable flow of electrical current in the electrical circuit of the playback head because the average level of flux in the gap won't vary. Different polarizations will appear side by side within the gap and will cancel one another out. If the playback head gap is not perfectly perpendicular to the tape, high frequency response will also be adversely affected, and just a couple of degrees of misalignment will be sufficient to eliminate response past 10kHz. And even if the gap is as narrow as is practical and is perfectly aligned, the cassette deck will require heavy equalization to achieve usable response out to 20kHz.

The perpendicular alignment of the playback head to the tape passing beneath it is known as azimuth alignment, and in many, perhaps most high quality tape decks, the head may be adjusted within it's mounting for absolutely correct azimuth alignment. Indeed really scrupulous home recordists check azimuth alignment at fairly regular intervals because even minor drift can exert such a major effect on top octave response. The importance of azimuth alignment has been publicized by a fair number of manufacturers, one of which has even developed an automatic azimuth adjustment feature for the playback head. While not disputing the significance of azimuth alignment, we believe that most manufacturers are employing half measures in this regard. It is one thing to set azimuth based on the assumed position of the tape in the tape path, and quite another to maintain correct azimuth alignment when the tape is in motion. The fact is that careful azimuth adjustment is to little avail unless the tape transport is rock steady and prevents any appreciable tape bounce.

At Harman Kardon we set azimuth correctly at the factory and provide precise adjustment for subsequent recalibration, but we also engineer our tape transports so that azimuth remains fixed under dynamic conditions of tape playback. And that way the listener gets all the highs every time.

Now let's look at some of the consequences of the standard 1 7/8" per second record speed specified for the cassette medium.

Because of the slow speed of cassette tape, the length of tape occupied by very high frequencies of over 10khz is mere microns—millionthes of a meter. Therefore the width of the playback head must be of comparable dimensions, and the difficulties in aligning such a tiny gap to within a fraction of a degree are considerable. In contrast, professional tape recorders with tape speeds seven, even fourteen times as fast, can spread the high frequencies out over nearly a millimeter and use much wider heads to pick them up. The wider headgaps need not be so critically aligned, and more

Harman Kardon TD4800 Cassette Deck

important they are much more efficient, producing larger electrical currents in the pickup head circuits and ultimately better signal to noise ratios. And the last point is important because the noise from cassette tape tends to be concentrated in the high frequencies, and thus lowering efficiency and decreasing the signal to noise ratio to get more highs tends to be self defeating because you end up getting more high frequency noise as a result. Dolby noise reduction is a way around the problem, but until the recent advent of Dolby S, it was a very imperfect solution (Fig. 6d).

You should note that a narrow gap width is only required for the playback head in an extended frequency range cassette deck, not for the recording head—in fact a fairly wide gap is preferable there. That's because the record head only records at the edge of the magnetic gap, the whole of the gaps is not utilized. But because wide gaps make for greater efficiency, a recording head gap will be made fairly wide—if it's to be used for recording only. Now on lower priced cassette decks the cost of separate record and playback heads becomes prohibitive, and one head has to serve both functions. Inevitably it's a compromise.

Obviously a head with a gap width of less than a micron is going to require manufacturing tolerances in the angstroms which is level of precision far exceeding Swiss watchmaking standards, though not beyond our capabilities here at Harman Kardon. Such precision is not to be had cheaply which is why cassette decks with extended high frequency response are not cheap either. But head dimensions are only one part of the equation. Equally important are the materials and construction techniques used to fabricate the heads.

Most current high quality

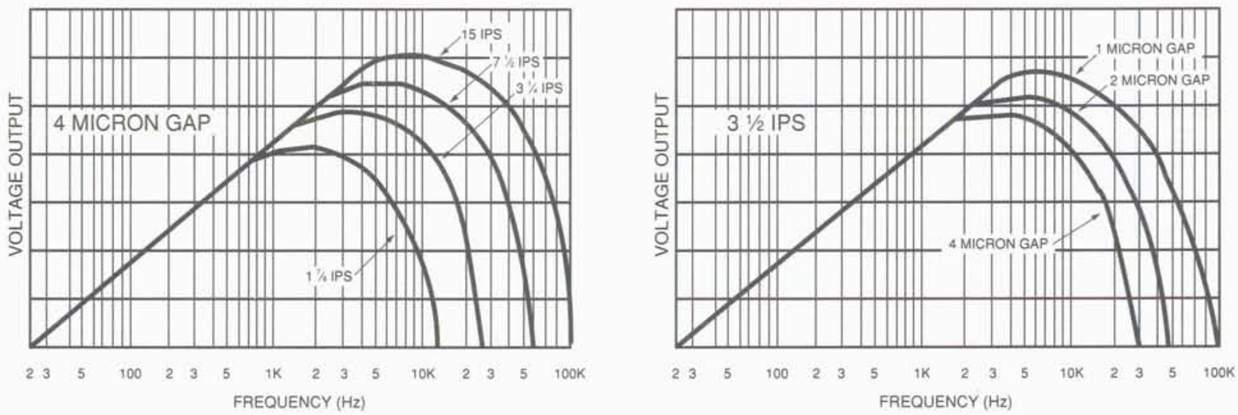



FIG. 6d

cassette heads are made of either permalloy, sendust, or so called isotrophic alloys—all of which are iron alloys with special magnetic properties. There are many different grades and varieties of each material, and none is ideal in all respects. We use isotrophic heads in our top decks and hard permalloy in the rest of our line. Both materials offer a good balance of high magnetic permeability (permeability refers to the ease with which a substance may be magnetized) extended high frequency response, high saturation, and long wearing qualities.

Whatever the material used for the tape head, controlling unit to unit variations in heads is absolutely critical, and here's why. All tape head exhibit falling response past 15kHz, and this droop in response must be compensated for by electronic equalization. The equalization itself is achieved by means of a narrow band peaking filter, and in order for such equalization to bring about the desired result, the head must be completely predictable and consistent. To meet this need Harman Kardon carefully tests all of the heads that are used in our decks. We accept only 15% of all the heads tested.

## A WORD ON DUAL WELL CASSETTE DECKS

One of the most successful components categories to appear in the last decade has been the dual well or dubbing cassette deck intended primarily for copying program material from one cassette to another, and secondarily for extended playback time in background music applications. Consumers in general have embraced the dubbing deck, but discriminating audiophiles have been overwhelmingly negative, and with good reason. Almost all dubbing decks offer poor fidelity. Our dubbing decks are the sole exceptions.

In setting about designing dubbing decks which would achieve the same 20kHz bandwidth as our single well decks we evaluated the engineering problems presented by the design and concluded that there was no compelling reason why a dubbing deck could perform at a very high level. The presence of auto-reverse proved the biggest obstacle because of the difficulties ensuring per azimuth alignment in both directions, but such difficulties are not insurmountable as proven but the existence of high quality single well decks with auto-reverse. We found that by simply applying our usual stringent standards of magnetic head and transport design to both wells and designing the component as essentially two separate decks in one chassis, we could provide our customers with essentially two separate high performance decks operated by common controls.

Harman Kardon dubbing decks use completely separate transport motors for either section, and separate high quality discrete signal circuitry. They are unequivocally the best sound dubbing decks in the world, and they're also the most user friendly due to the powerful proprietary microprocessor control system we developed specifically for this application.

## TAPE EQUALIZATION AND THE EFFECTS OF LOW LEVEL AMPLIFICATION

Magnetic tape is a highly nonlinear recording medium, as we said before, and in the absence of electronic preconditioning of the signal, a magnetic recording will be very distorted and far from flat in frequency response. The frequency response problems are most pronounced at low tape speeds, and the cassette, as we've seen is the slowest speed format around. Even with a perfectly aligned, high quality recording head, response starts to droop past 2kHz. and it's way down by 10kHz.

To counteract this droop, the tape is equalized during the recording process (Fig. 6e). Note that the way that equalization is used here is fundamentally unlike what occurs in the RIAA networks used for phonograph disc recording and playback. The phonograph record is an essentially linear recording medium, and equalization is used to increase headroom and reduce noise, and the record equalization is exactly reversed on playback to restore flat response. In the cassette medium equalization is used to restore response both in the bass and in the high frequencies, and a lot of it is required. It is not reversed on playback.

Because of difficulties in obtaining extended frequency response in a cassette recording, frequency range has become a

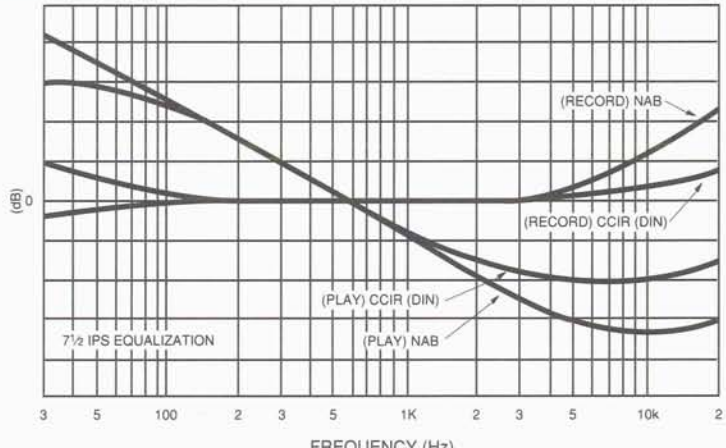


FIG. 6e

focus of attention and an arena where "spec wars" may be waged. Record equalization has become a weapon in these spec wars.

Due to problems in tape saturation when a cassette recording is heavily equalized at normal recording levels, manufacturers, when testing for spec, have taken to bringing the average recording level down some 20dB from the 0dB level mentioned earlier. The 0dB level is conventionally stated to be the level at which tape distortion reaches 3%. This is worthy of note because professional reel to reel tape recorders are commonly specified at only 10dB below the reference level. A 10dB reduction in input level has a big impact on the signal to noise level on the recording.

Why do most manufacturers measure down at -20dB? Because tape saturation simply won't permit them to get extended response at the higher recording level. So they use a level which is lower than anyone would ordinarily use—one that's not real world. They fudge to make spec, and a deck specced for 20kHz response at -20dB may not achieve anywhere near that high frequency response at -10dB. Ours will, and in fact all Harman Kardon decks are specified for flat response past 20kHz at 0dB.

A note about equalizing for high frequency response: to put a lot of boost in a signal you, need a lot of gain in the amplifier section of the tape deck which tends to raise the noise floor and increase distortion. If you're using an extended frequency response playback head of low efficiency, you'll need even more gain. And this is an area where Harman Kardon's meticulously engineered all discrete tape deck electronics really shines. The excellent line amplifiers used in our tape recorders simply permit a cleaner signal to be recorded onto the tape.

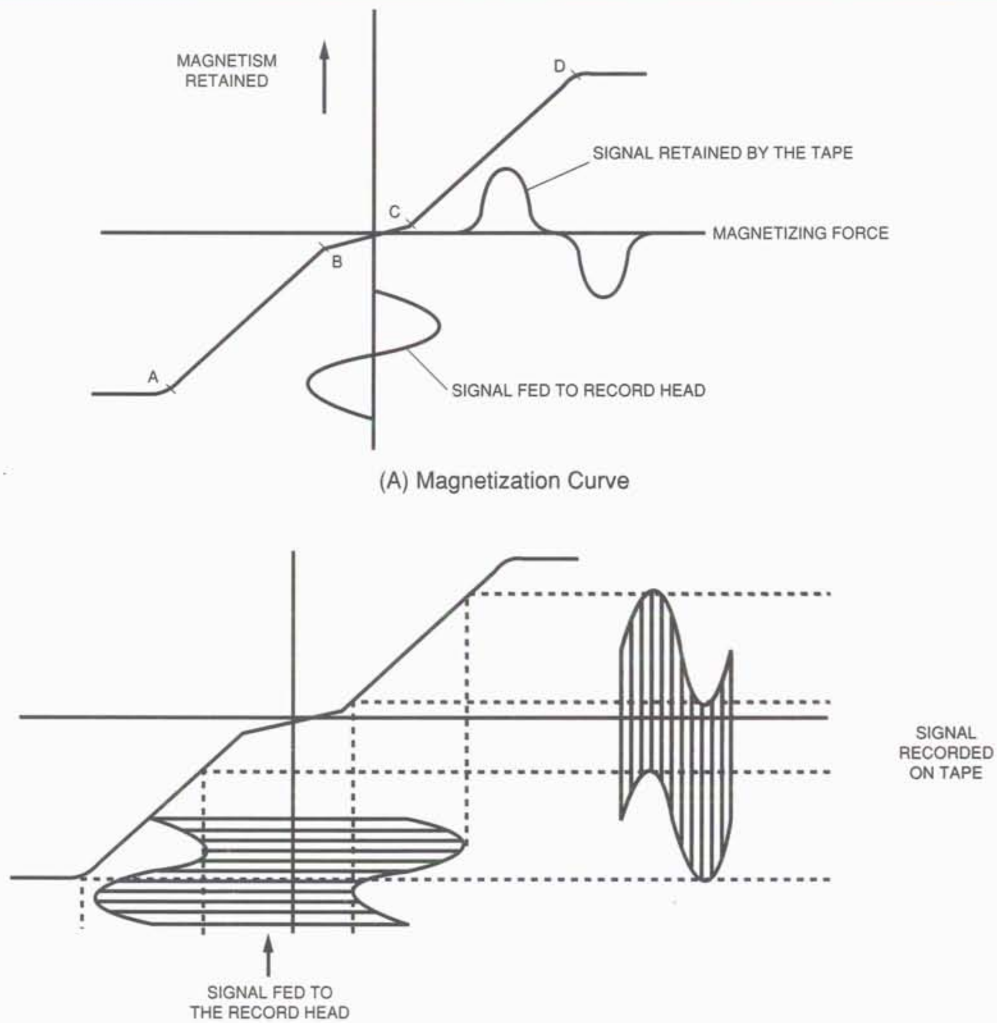
## TAPE BIAS AND DOLBY HX PRO

We've already mentioned the fact that magnetic tape is a high distortion storage medium. But that's not to say that it can't be linearized. This is done by means of a bias signal during the recording process.

A bias is a constant signal which is used to ensure that the tape is always partially magnetized. That's because the tape itself has a certain resistance to magnetization called reluctance that has to be overcome before a signal can be impressed upon it. The inertial effects of reluctance are manifested in distortion during playback, but they can be avoided by feeding the tape a constant signal of low magnitude. To avoid creating background noise that low magnitude bias signal is placed at around 100kHz, far beyond the range of audibility and hopefully where it won't modulate with the music signal.

In actual fact the bias frequency selected by most manufactures is too low for a cassette deck with wide frequency response. Most decks made today have the bias set at 85kHz. That's low enough to create intermodulation distortion in the audio band when response goes out to 20kHz.

True 20kHz and 85kHz frequencies won't generate audible modulation frequencies by themselves, but a clipped 20kHz signal will create plenty of harmonics extending up to 70kHz.




(A) Magnetization Curve



(B) After Bias

Illustrating effect of biasing on recording linearity.

FIG. 6f

and beyond. Subtracted from the 85kHz bias, these will cause modulation distortion products from 5kHz up to 20kHz. For this reason Harman Kardon decks use a 105kHz bias frequency.

Bias level as opposed to bias frequency will vary for different types of tape, and all high fidelity cassette decks made today can be calibrated for common tape types—in fact many, including a number of our models—calibrate themselves automatically. This is important because the bias setting has a significant effect on sound quality. Underbiasing results in excessive distortion, while overbiasing robs the tape of headroom and leads to premature tape saturation.

During the first years in which cassette was establishing itself as a high fidelity medium, designers of high performance decks began to consider a problem in the biasing which had existed as long as bias signals had been used with magnetic recorders, but had not been considered critical in regard to high speed reel to reel machines. It was observed that at high frequencies the music signal itself would combine with the bias signal and overbias the tape, leading to high frequency saturation problems.

In 1980 Dolby Laboratories developed a dynamic biasing system called Dolby HX which automatically reduced bias level during recording in the presence of high frequency information for better high frequency headroom. Shortly thereafter an improved version of the system called HX Pro was developed. Harman Kardon was one of the very first manufacturers to utilize either system in a consumer product, and today we include HX Pro in virtually all of our cassette decks.

## DOLBY B, C, AND S

Dolby Labs developed the first widely accepted noise reduction system for magnetic tape, and today Dolby virtually owns the market for noise reduction in both professional and consumer applications. At Harman Kardon we have long enjoyed a close business relationship with Dolby Labs and we were the first manufacturer to introduce three key Dolby technologies to the public, namely Dolby B, Dolby HX, and Dolby S.

Dolby's noise reduction systems work on a principle known as compansion. Dolby did not invent compansion, but in our opinion the company has used the principle more intelligently than its competitors.

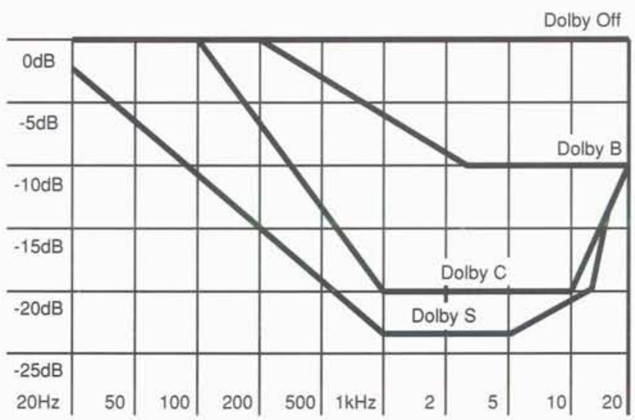
Compansion is short for compression/expansion. In a compander system, the voltage swing of the signal is compressed during recording, and expanded during playback. The compression, by limiting the negative voltage swing, brings the signal well above the noise floor of the tape, while keeping the positive swings from saturating the tape. By expanding the signal on playback through a decoder you restore full dynamic range electronically while retaining the advantage of a lowered noise floor—at least most of the time.

Compansion in its basic form will definitely improve signal to noise ratio in a tape recorder, but as with so many other engineering there's no free lunch. The problem with simple one band compansion is that you will clearly hear noise getting louder and softer in the presence of strong bass and in the absence of treble sounds to mask the noise. Such systems are said to "pump and breathe", and they also suffer from subtler modulation distortions arising from the expansion process.

Dolby's initial system, called Dolby A, addressed these problems by applying compansion separately and differentially over four bands. Any out of band noise created by the expansion process within one of the bands would simply be rejected by the band filters. Of course there's quite a bit more going on in Dolby A than just band splitting, and the fact that the system is still commonly used today twenty-five years after its introduction attests to its effectiveness.

Dolby A provided a maximum of 15dB of noise reduction, and it generally worked admirably, but the circuitry required for encoding and decoding was much too complex and expensive to lend itself to consumer applications. So in 1970 Dolby announced a much simplified version of the system developed specifically for the cassette format. It was called Dolby B, and it's still very widely used today. As indicated earlier, Harman Kardon was the first manufacturer to embrace the system.

The Dolby B noise reduction systems has only one band of compansion, and that band is restricted to the higher frequencies (Fig. 6g). What makes the system unusual and far more effective than other single band companders is the fact that the corner frequency of the compansion band is designed to slide up and down continuously according to the tonal balance of the program material. In the presence of a lot of upper midrange energy, the band will slide up so that noise reduction is restricted to the extreme highs, and thus pumping and breathing effects will not occur, but where




Relative Noise Reduction Effects

FIG. 6g

middle frequency signals are lower in level, the filter will slide down to provide noise reduction over a wider range.

Dolby B offers at most only 10dB of noise reduction but that's concentrated above 4kHz where noise is most audible and so the effect of Dolby B is fairly dramatic.

Dolby C came later, in 1980, and improved the noise reduction effect to 20dB for frequencies above 1kHz, a very significant improvement. Like Dolby B, Dolby C has only one sliding band, but the band itself has two stages of compansion which work on different signal levels, and which permits more compression to be used without audible problems. Today Dolby C is offered in the majority of high quality cassette decks, though it has never become standard in the way that Dolby B has, nor has it ever been much used in prerecorded software. As was the case with Dolby A, Harman Kardon adopted the new technology virtually as soon as licenses were issued, and today we include it across most of our line.

In 1985 Dolby announced a new professional noise and distortion reduction system called Spectral Recording—usually shortened to Dolby SR—as a replacement for Dolby A. The improvement was very significant, and Dolby SR provided dynamic range of over 100dB at the recommended recording speed of 15 inches per second, exceeding professional digital standards. Acceptance by the recording industry has been very enthusiastic, and today most recordings are mastered in analog on Dolby SR reel to reel tape decks.

Unfortunately the complexity of an SR module is an order of magnitude greater than a Dolby A processor, and the circuit contains over 1500 transistors, capacitors, and resistors. A single channel Dolby decoder is almost as big as stereo tuner even with integrated circuitry. Still the principles behind Dolby SR proved too promising to be ignored in the consumer sphere, and Dolby Labs set about developing a simplified version of the system that would bring about similar improvements in the cassette medium. The new system was announced a year ago, and it's called Dolby S.

Dolby S is much closer to its parent SR than Dolby B is to Dolby A.

It would take a whole chapter to describe all of the signal processing functions performed by a Dolby S module, but the basic principles of the system are simple enough. Dolby S noise reduction is concentrated in the range of audible frequencies above 400Hz, or roughly the top half of the audio spectrum, though some noise reduction is provided in the bass as well. Two overlapping bands are used confine the compansion action to selected frequencies; one of these bands is fixed and extends from 400Hz to 12.8kHz while the other has a high pass filter that slides from 200Hz to past 10kHz. A single fixed noise reduction band takes over below 200Hz.

The action of the fixed and sliding high frequency bands working together ensures that all frequencies get the treatment they need for lowest noise throughout. Intense signals are limited, while low level signals are boosted with no significant gaps in the spectrum. With Dolby B and C, frequencies below the dominant frequency receive no noise reduction while in Dolby S they are treated.

Dolby S provides up to 24dB of noise reduction above 400Hz and 10dB below that point. It is the most effective noise reduction system yet developed for a consumer format.

At Harman Kardon we are proud to have been the first manufacturer to feature the new system in our cassette recorders.

## SUMMARY

Harman Kardon cassette decks have been designed to offer the finest values in their product category. Every aspect of component design has been critically assessed and optimized, including head construction, transport mechanics, and tape electronics. We believe that our top of the line decks equipped with the new Dolby S system are the best sounding recording devices yet offered to the public, and we will match the frequency response of our lower priced units against that of any deck from our competitors at any price.

7

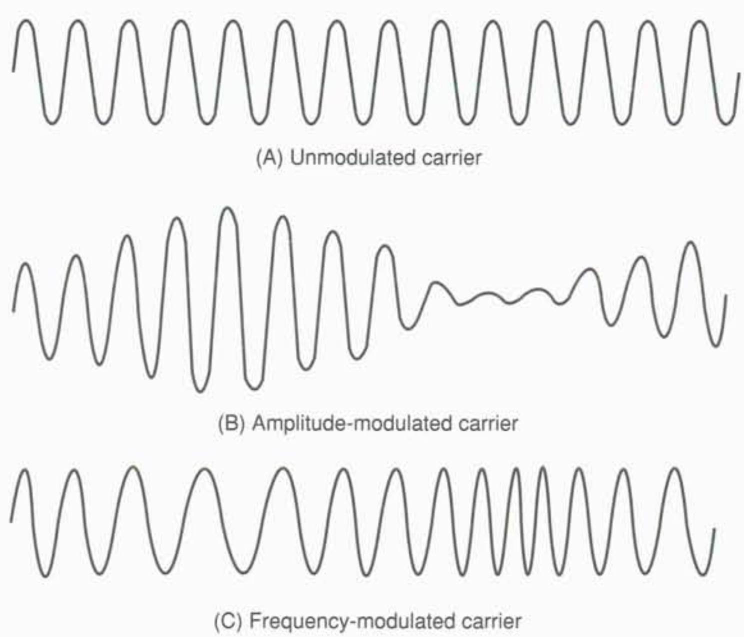
SECTION 7

# AM/FM STEREO TUNERS

Harman Kardon has always had a very strong commitment to providing superior radio broadcast reception, particularly in regard to high fidelity stereo FM programming. We must acknowledge that FM broadcast reception does not now receive quite the emphasis it once did in high fidelity circles, but the fact remains that a pristine, uncompressed live FM broadcast heard over a high quality tuner can provide a level of realism which is difficult to match with recorded program material. A tuner remains an important component of a home entertainment system, and in this category, as in all others which we manufacture, we attempt to provide our customers with the most advanced technology at a fair and reasonable price.

## THE FM DILEMMA, OR WHY ALL AROUND EXCELLENCE IS SO DIFFICULT TO ATTAIN IN A HIGH FIDELITY TUNER

The designer of high performance stereo tuners must strive to meet two principal design goals, which, so long as conventional technology is employed, are not entirely compatible. First the designer must configure circuitry which will exhibit generally exemplary behavior in passing an audio signal—that is, will maintain low noise, low distortion of all sorts, good phase linearity, and adequate dynamic and frequency response capabilities. At the same time, the




Comparison of amplitude and frequency modulation

FIG. 7a

designer must create a circuit which will discriminate between a multitude of signals located very close together in frequency, and reject interference from undesired signals even when the desired signal is relatively very weak.

A very brief look at how FM operates will indicate the (Fig. 7a) difficulties of meeting both goals.

FM, as you're undoubtedly aware, stands for frequency modulation. What this means is that the transmitted signal consists of a high frequency carrier whose exact frequency varies according to both the frequency and the amplitude of an audio signal which is impressed upon it.

Reception of an FM signal involves a process of detecting and isolating the relatively narrow frequency band where the signal is located, and the only practical way to accomplish that is to use tuned resonant circuits commonly made up of electrical filters of one sort or another. The effectiveness of these tuning circuits in selecting the desired signal and rejecting all others is a function of the sharpness of the electrical filters used in the tuned resonant circuits. Electrical filters by their very nature engender phase shift which is always manifested within the frequency range of interest, and furthermore, the sharper and more effective the filters, generally the greater the phase shift. Unfortunately, frequency modulated signals are highly sensitive to phase shift, which in turn is manifested in distortion products in the demodulated audio signal. In other words the sharper the filter, the more effective it will be in selecting the desired signal and suppressing interference, and yet the more distortion it will generate in the process. What's worse, the phase shift will also tend to reduce stereo separation in as much as the subcarrier bearing the stereo information must bear a precise phase relationship with the mono signal.

So the very act of detecting the FM broadcast signal is going to introduce some distortion into the signal, and the greater the ability of the tuner to lock onto a given signal and reject all others, the more distorted demodulated audio signal is likely to be. It seems you just can't win.

## INTERFERENCE AND FM BROADCAST RECEPTION

Lest the preceding discussion be taken to indicate that sharp filters and highly tuned detector circuits are completely undesirable, let us hasten to add that without the ability to discriminate among broadcast signals with a high degree of resolution, an FM tuner is practically worthless. The FM bands are simply too crowded, and there's too much stray radio frequency energy in the environment for an FM tuner to function without high selectivity. Gentle filters might be permissible in an area with only a handful of stations, but throughout most of the United States that's not the case. Throwing away selectivity and resistance to interference is simply not the answer, in fact our position at Harman Kardon is that most FM tuners aren't nearly selective enough.

To understand our reasoning in this area, it helps to have a basic understanding of the forms of interference with which a tuner must cope.

Interference in the FM band comes from two principle sources, from strong broadcast signals at frequencies near to those of the desired channel, and from multipath reflections from the desired signal itself. Both problems are significant, though the first tends to be much more intractable.

Let's consider the first problem for a moment.

In assigning frequencies to FM broadcasters, the Federal Communications Commission has broken up the country into a multitude of broadcast areas, and within each area, broadcasters are only assigned alternate broadcast channels—in other words, half of the channels are left unoccupied. That's because FM works by spreading the carrier frequency out into sidebands of nearby frequencies; the channels have to be widely separated or they will tend to overlap one another.

The general system of FM frequency allocation works well enough in most places, but in boundary areas between two crowded FM markets it can break down, particularly in areas where bodies of water or unoccupied flatlands permit FM signals to travel for long distances. In such cases a signal may be subject to a degree of interference from an adjacent channel in another broadcast area that is sufficient to degrade reception severely. This is known as adjacent channel interference.

Multipath distortion, the second significant type of interference, is a very different phenomenon which arises when a given signal is reflected from some very large physical object such as an office building or a hillside, and the reflection then merges with a the direct signal thereby creating a series of cancellations and reinforcements. Here the desired signal is in effect interfering with itself, and multipath may manifest itself when no other signal is present. The effect of multipath distortion on reception is highly dependent upon the position of the receiving antenna, and frequently by moving the antenna as little as couple of feet, the listener can eliminate the problem. Multipath tends to pose more of an annoyance in automotive tuners whose antennas are constantly moving than in domestic audio systems, but in some areas it can seriously interfere with reception in stationary installations as well.

## OVERCOMING INTERFERENCE - OLD VERSUS NEW TECHNOLOGY

Both adjacent channel interference and multipath distortion are primarily problems of FM radio. AM has its own set of problems, chief among them a very low immunity to radiated electrical noise, but the channels don't interfere with one another because the carrier frequency isn't spread, and multipath seldom arises because AM's enormously long radio waves are not reflected by buildings or hills. But FM, with its relatively short waves and wide channel bands, is inherently vulnerable to both of these problems.

As we have seen, the obvious way to improve selectivity is to tighten filters and sacrifice linearity and stereo separation. Interestingly, the same obvious solution is applicable to multipath problems because the multipath reflection, due to Doppler distortion, varies slightly in frequency from the direct signal.

Most manufacturers with a concern for sound quality and signal integrity have not chosen to tighten filters (nor in fact has Harman Kardon—at least not in the usual sense). The typical audiophile solution to interference problems is to equip the tuner with either a "stereo blend" adjustment, or a special narrow band mode with variable filtering as a selectable operating parameter. Both solutions permit high fidelity reproduction in the absence of interference, while allowing the user to forgo linearity if so desired in order to suppress strong interference.

At Harman Kardon we prefer to have our cake and eat it too. We want our customers to enjoy low distortion and wide stereo separation regardless of the r.f.


Harman Kardon TU930 AM/FM Stereo Tuner

environment, and we've provided the means for them to do so.

## CONSISTENTLY SUPERIOR FM RECEPTION

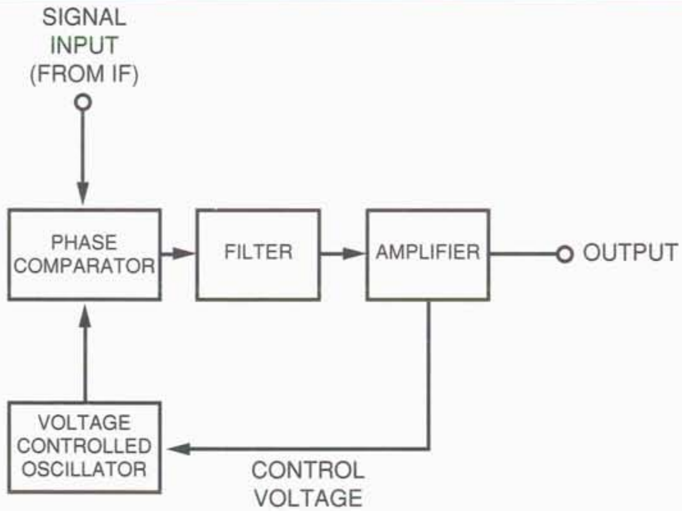
Harman Kardon pursues superior FM reception through three principle design strategies. The first is basically business as usual—discrete circuits in the buffer amp stages, proper grounding, generous power supplies, and careful circuit board layout. The second strategy involves a type of IF circuit unique to consumer audio called Active Tracking. And the third is known as digital fine tuning.

We won't discuss the first strategy in any detail here. Our rationale for using discrete circuitry and ground isolation has been discussed elsewhere. The other two strategies require some explanation though.

Active Tracking constitutes a special use of phase locked loop frequency synthesis, a technique whose application in consumer tuners was pioneered by Harman Kardon.

A phase locked loop is a type of servo or feedback oscillator circuit (Fig. 7b) which will generate a precise frequency with a very high degree of accuracy and freedom from drift. It has many uses in presentday electronics of which radio frequency tuning is one of the more common.

Phase locked loop tuning circuits go back to the early thirties and have long had a place in demanding military radio




Phase locked loop.

FIG. 7b

applications, but their use in consumer audio—which we at Harman Kardon pioneered—is only about twenty years old.

In a typical phase locked loop circuit, a quartz oscillator provides a highly accurate reference frequency on a continuous basis. This is compared to the output of a voltage control oscillator (VCO) whose output is a function of the voltage at its input. Any difference between the output of the VCO and the reference will result in a DC offset voltage which changes the frequency of the VCO output until it matches that of the reference oscillator. In total the phase locked loop circuit constitutes a classic feedback arrangement.

In order to use this circuit to select a frequency, one feeds a wideband input from the antenna into the input of the VCO which responds strongly only to those frequencies near to the frequency generated by the reference oscillator. The feedback loop is adjusted to permit just enough moment to moment drift to register the modulations of the FM carrier and no more. The circuit is very highly tuned, but uses no conventional electrical filter elements to select the frequencies, that is, no capacitors or inductors. And it works beautifully, so beautifully in fact that nearly every tuner made today uses a phase locked loop in its front end RF tuning section.

In circumstances of low adjacent channel interference, the PLL detector is normally all you need to obtain a good clean signal, but in spite of the circuit's superior discrimination it can't suppress a really strong adjacent channel signal—at least not by itself. However if you add another stage of PLL tuning at the IF (intermediate frequency) stage of the tuner which comes after the detected signal is converted to another carrier frequency called the IF, then the filtering effect is multiplied. This is precisely what Harman Kardon does in the Active Tracking circuit.

Before we examine this second PLL stage in detail, it might be well to review a few FM fundamentals. All consumer tuners made today use a tuning technique called heterodyning—also known as superheterodyne. The object of heterodyning is to convert every FM signal received to the same carrier frequency. That way the demodulator section which removes the carrier does not have to be separately tuned for each station, and tuning need only take place in the front end of receiver. The one universal frequency to which all carriers are converted is called the IF or intermediate frequency, and it is generally set at 10.7MHz.

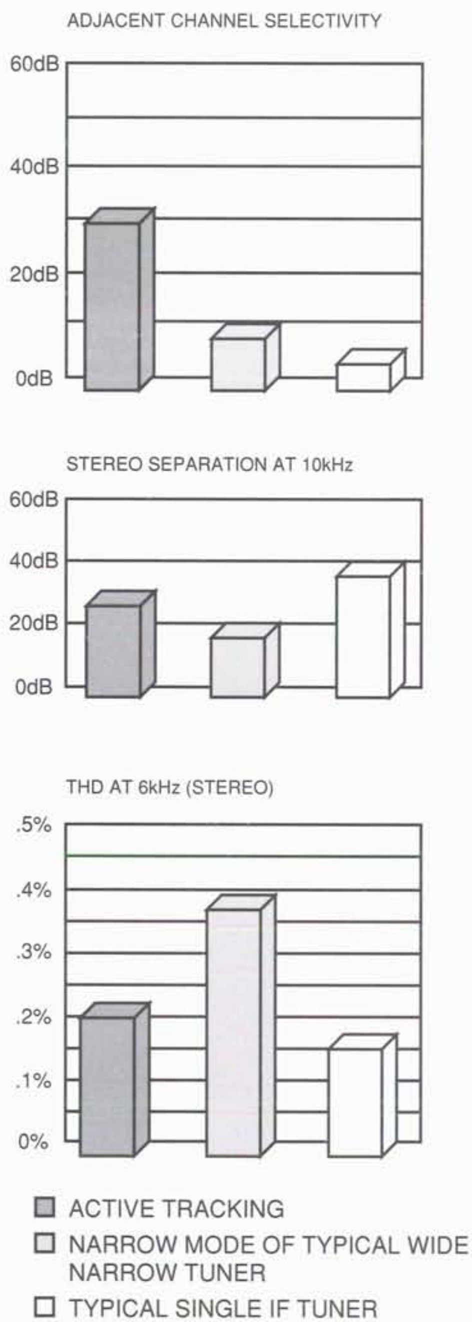



FIG. 7c

Most high quality tuners have a special filter in the IF section for suppressing adjacent and alternate channel interference, and these filters tend to be effective as suppressors only to the extent that they shift phase within the desired channel band. Early IF filters were made up of common filter elements that is capacitors and resistors, but for the last twenty years or so, special filters made of piezoelectric devices have predominated. Both types have their problems, and although IF filters have been the subject of an enormous amount of design work, nothing has proven completely satisfactory.

When a PLL tuning circuit is substituted for an electrical filter within the IF section, a real performance breakthrough becomes apparent (**Fig. 7c**). The phase locked loop rejects any frequencies outside its narrowly defined window, and it does so without engendering phase shift.

The result of this extra phase locked loop is that the signal will maintain excellent stereo separation and signal-to-noise in the presence of adjacent channel interference where typical tuners will produce a monophonic signal distortion often exceeding 10%. And that's a unique benefit. And it's a benefit only available in Harman Kardon tuners having the Active Tracking circuitry.

## DIGITAL FINE TUNING

PLL frequency synthesis tuning is probably the single biggest improvement in stereo tuning technology devised since the advent of stereo FM, and in the face of PLL the old variable capacitor detectors that preceded it disappeared virtually overnight. Today hardly anyone seriously disputes the overall superiority of PLL, however there is one area where typical PLL detectors do not compare favorably with the old technology, and that is the area of fine tuning.

In theory a PLL circuit can be tuned precisely to any frequency one wishes, but in normal consumer implementations of the circuit, the actual tuning is performed by a digital microprocessor that only permits the reference oscillator to operate at specified frequencies corresponding to the standard FM channel allocations. And in one sense that's the strength of the system because it frees the user from having to search for the station. The designated frequency is tuned exactly and it stays tuned indefinitely. There's no fine tuning at all, just correct tuning.

More often than not this approach works just fine, but in some instances broadcasters themselves detune their transmitters slightly in order to overmodulate their signals a little more and get just a bit more signal strength. Moreover in areas of severe multipath a slightly off frequency reflection may actually provide a best signal. So right on the money doesn't always make for the best reception.

To deal with such situations we've provided our tuners with a microprocessor that can tune in increments that are small enough capture any detuned signal. Other manufacturers have developed circuits that can tune halfway between adjacent channel slots, or utilize auxiliary variable capacitor tuning circuits, but ours retains the precision of PLL while approximating the infinite resolution of the variable capacitor type. It's just another case of having our cake and eating it too.

## AGAIN GREATER VALUE

All Harman Kardon tuners, even those that are part of receivers, use discrete buffer amps and well isolated properly engineered power supplies. All, even the least expensive, are consistently musical sounding.

Digital fine tuning appears in are more expensive models, and Active Tracking is presently confined to our flagship.

FM tuner design is an area where we perform ongoing research and where we aim to maintain a position of dominance in design expertise. Our tuners have always borne comparison to far costlier models from other manufacturers, and have frequently been selected as broadcast monitors by radio stations with a demonstrable concern for sound quality. We will continue to seek ways to improve FM reception, and we will continue to set the standards for affordable stereo tuners.

8

S E C T I O N 8

# HOME THEATER

This chapter is a little different than what has come before because we will be focusing on system concepts as much as on specific products. It's to your advantage to read this material very carefully as the concepts discussed here have become critically important in the retailing of audio and video components.

## HOME THEATER - ATTEMPTING A DEFINITION

"Home theater" has achieved the status of an industry buzz word, a term invoked by people at every level of the industry and in all market segments. In fact, it's safe to say that no idea has generated as much excitement in the industry since digital in the early eighties and stereo back in the late nineteen fifties.

Home theater really is exciting, as products in this category have caught the attention of consumers, and represent the area of greatest growth within the audio/video marketplace. Here we'll discuss how you can build on that excitement and interest.

But first a definition.

In the broadest sense home theater signifies a room in which the viewing and listening environment is similar to what you would encounter in a well equipped, properly calibrated movie theater. Achieving that requires five basic elements:

1. Video software, with audio tracks recorded in Dolby Surround or another surround format.

2. A laserdisc player, Hi-fi VHS tape deck, satellite receiver, or stereo television set to reproduce the software (some Dolby encoded material is now broadcast by local television stations or by satellite.

3. A large screen television—either a 27" (diagonal) or larger direct view monitor, or a projection system.

4. A stereo/surround sound audio system.

5. A suitable listening/viewing space.

A complete home theater system, with left, right and center channel speakers, a large screen TV and a powered subwoofer.

Obviously there are a lot of A/V packages optimistically designated as "home theater systems" by manufacturers or retailers which do not include all of these elements. They don't provide the full movie theater experience and are merely trading off the catch phrase of home theater.

At Harman Kardon we want nothing to do with such deceptions. On the contrary, we believe that the home theater customer is not likely to be misled, and that high performance is a must in A/V systems marketed as home theater. That's because virtually everyone has experienced sounds and images in a movie theater, and is in a position to make evaluations of the consumer equivalents. So if you want to succeed in this market, the systems you assemble and sell must deliver quality audio performance.

## THE MOVIE THEATER MODEL

To recreate a movie theater environment in a home setting, you first have to know a little about how a real commercial movie theater is designed, and especially how sound is delivered.

### The Well Appointed Picture Palace

A movie theater designer seeks to draw the audience into the narrative action of the film with a large, bright image that fills the field of vision. Sounds will follow the on-screen action and, at the same time, seem to extend out from the screen and around the viewers. Indeed, in modern film making, sound is a key element in giving the film a three dimensional aspect.

Now let's consider the equipment required to provide that experience in a movie theater, with particular emphasis on the equivalent components in the home setting.

Obviously the film projector is the central component in the motion picture playback process because it normally forms the signal source for both sound and picture. For the purposes of this discussion we can ignore the film projector, since its place is taken by video transmission which works on entirely different principles. Instead we should turn our attention to the screen, the only optical component that a movie theater and a home theater have in common.

The screen in a movie theater—or a home theater—is the anchor for the whole entertainment experience. It displays the images comprising the motion picture, and provides a spatial reference for the localized sounds on the audio track. This last point is very important, and you must grasp it fully to make any sense of the discussion to follow.

Modern movie soundtracks follow the action on screen. If a car moves from the left side of the screen to the right, the car's sound will seem to follow. If someone is facing the camera and speaking, the sound will seem to come from the center of the screen. If a helicopter approaches from the distance, and then appears directly overhead, the sound will seem to come out of the distance and then to pass right over the head of the audience. In some cases sounds will be non-localized, as when the scene takes place in a forest and noises are heard all around.

All of these effects are routinely handled by a modern movie theater sound system, and they can also be reproduced in a properly assembled home theater system. In fact, if they're not correctly reproduced, the customer is being shortchanged.

Movie theaters achieve these effects of sound placement by extracting at least four channels of sound from the stereo soundtrack, and by routing those four separate signals to a carefully placed array of loudspeakers. The process required to do this is a bit involved, but it's well worth studying, because it is performed in almost exactly the same way in the home, and because unless you understand it, you are unlikely to be able to achieve consistent results in setting up systems.

**Multi-channel Motion Picture Soundtracks**

Modern motion picture soundtracks typically begin as multi-channel recordings. In other words, there are four or more channels rather than simply a stereo pair. In most movies the multiple channels are matrixed or encoded into a stereo pair. On playback the extra channels are decoded and appear highly separated from one another. This means that one particular sound may be confined to a single channel and will not be heard from the other channels. The method for combining multiple channels into two is called matrix encoding, while the opposite process of extracting them is called decoding. We'll have much more to say about this later, but for now just keep in mind that movie sound is essentially multi-channel sound.

Now let's identify these multiple channels.

Movie soundtracks include the same basic left and right channels found in any stereo recording, and these are normally reproduced by a pair of speakers at the left and right sides of the screen. In addition to the stereo left and right, there is a center channel which is normally reproduced through a speaker located behind the center of the screen. Finally, in the most basic multi-channel theater setup, there is a fourth channel


FIG. 8a

In post production, the soundtrack channels are encoded onto two audio tracks.

During film presentations in the theater, the two audio tracks are decoded into the four channels of the original soundtrack.

whose output is usually distributed into an array of speakers positioned around the sides and back of the theater.

A four channel arrangement is standard for playing back what are known as Dolby Stereo™ optical soundtracks, by far the most common format for movie sound today (Fig. 8a). These Dolby Stereo™ optical recordings consist of a photograph of the soundtrack which is printed onto the film. It contains the two tracks of encoded audio information that are decoded into four channels when the film is shown (Fig. 8b).

In addition to the standard Dolby stereo optical process, there are several six track and one eight track format in use today, all of these having fully discrete sound. (Since these are discrete systems, no matrix decoder is used to reproduce any of these six and eight track formats.)

Until very recently six or eight channel sound was provided only with big budget, wide screen 70mm prints shown in select theaters.

DIGITAL dts™ SOUND
IN SELECTED THEATERS

DOLBY STEREO®
DIGITAL

SDDS
Sony Dynamic Digital Sound

FIG. 8b

All of that has changed, however, with the introduction of multi-channel discrete digital sound systems for theatrical use. The availability of Dolby Digital Surround DTS (Digital Theater Systems) and Sony's SDDS (Sony Dynamic Digital Sound) has brought the realism of discrete multi-channel sound to literally thousands of movie theaters around the world.

One note of semantics is in order here. You may see these systems referred to as "5.1" or "7.1" channel systems, even though they do have six or eight channels. In the 5.1 channel systems (Dolby Digital and DTS), there are separate front left, front right, center, rear left and rear right channels, with an additional channel for bass frequencies. The 7.1 system (SDDS) there are additional mid-front left and mid-front right channels.

So what is the ".1" channel? Easy, it's the bass channel. Although the bass occupies a fully separate and discrete channel, it is intentionally not a full bandwidth channel. For that reason it has been given the ".1" designation.

With theater chain and movie studio ads heralding these new advances in theater sound, your customers will have an even greater appetite for theater sound in their homes.

In the future we can expect to see greater penetration of digital sound in theaters, and variations of the digital coding systems will eventually bring them home via LV discs, HDTV, cable and perhaps even new videocassette formats. For the time being, however, the market is centered around matrix surround systems such as Dolby Surround, so we will confine the rest of our discussion to the techniques used to deliver sound in a typical home theater.

Now, let's examine the function of the four channels that heighten the listening experience in a home theater system.

**The Center Channel**

The center channel is critical since it is used for reproducing most of a program's dialogue. Most film dialogue is recorded in mono, and the center speaker is used to reproduce it because the sound will appear to be centered behind the screen at every listening position in the theater or home.

**Stereo Left and Right**

The stereo left and right channels are commonly used for a variety of purposes such as the music score or strongly localized background sounds such as footsteps. There's nothing rigid about the assignment of sounds to the front channels, and very often some of the musical information and background sound will be assigned to the center channel.

The most dramatic use that is made of the stereo left and right channels is an effect known as lateral panning where a sound source seems to sweep across the front of the listening space. Earlier we mentioned the example of a car moving across the screen and the sound of its engine appearing to follow it. This effect is created by first having a sound appear isolated in one of the stereo channels—let us say the left—then have it appear at a lower level in the center while still coming from the left. The sound subsequently fades in the left channel while getting louder in the center and then for a split second is present only in the center. Then it appears in the right as well while growing fainter in the center even as the right channel becomes louder. Finally it is present only in the right. The rapid change in amplitude relationships from channel to channel causes the sound to appear to move.

This process of lateral panning, while occasionally used in stereo music recordings, is much more appropriate as an accompaniment to onscreen action, and it has come to be one of the most commonly used effects of movie sound recording engineers.

With a complete quality home theater system, this effect can be enjoyed in your client's home.

**The Surround Channel**

In a theater, the surround channel speakers are distributed around the audience instead of behind the screen. They are generally used for low level, non-directional, ambient background sounds that appear to come from all around rather than one particular location. We already mentioned the example of forest sounds, but the surround channel will do just as well for reproducing street noise, the sound of a crowd, the wind blowing through a field, and so on. Many film directors also make use of the surround channel in addition to stereo left and right for background music scoring.

Occasionally the surround channel is used along with front center to create pans in depth where the sound begins in the front speaker and then is steered back into the surrounds or vice versa. Such a pan was used to accompany the image of the approaching space ship at the beginning of Star Wars, one of the first films to use the Dolby Surround

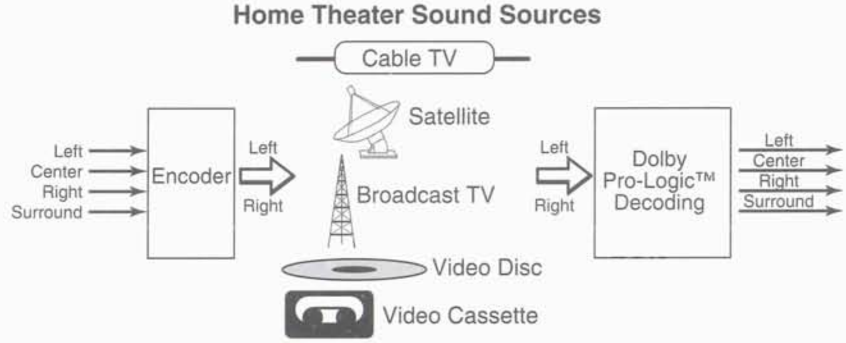



FIG. 8c
Home theater reproduction, like that in the theater, involves the decoding of the two audio tracks into the four channels of the original soundtrack.

process. This effect can be reproduced in a home setting, but that requires careful installation.

## TRANSPLANTING THE THEATER INTO THE HOME

Today's consumer video software formats and advanced audio-video playback hardware allow for a very close approximation of movie theater sound (Fig. 8c). Laserdiscs and VHS-Hi-fi tapes and a growing number of broadcast and cable programs, sports telecasts such as the NFL, and special events use exactly the same Surround encoding matrix as commercial film releases, and the better consumer surround sound decoders closely resemble the decoders used in movie theaters. In fact a Laserdisc digital audio track will actually provide higher fidelity than a typical Dolby optical sound track printed on film. Thus, with a properly assembled home theater system, you can provide your customers with a level of movie sound reproduction which may be as good or better than their local cinema.

## The Home Theater System

A home theater system, at its most basic, requires a stereo video source and a video monitor, three speakers across the front of the room, two surround speakers for the back of the room, a surround sound decoder, and a separate channel of amplification for each speaker. The amplifier channels and decoder may be combined in a single unit called an A/V receiver, though in the more expensive systems the decoder is typically separate from the amplifier.

An ultimate home theater system should also include one or more subwoofers with separate amplification. High-end systems tend to use projection television rather than direct view monitors, with front projectors being preferred.

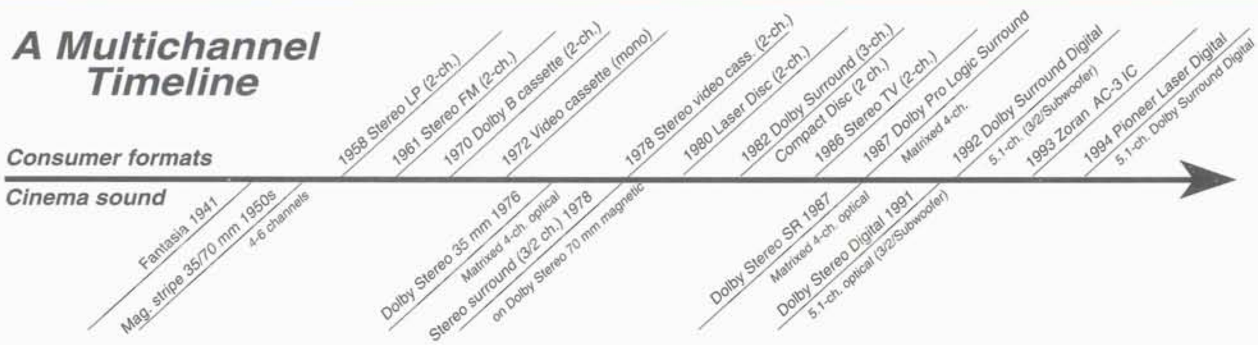
It is possible to put together a home theater system using ordinary audio amplifiers and loudspeakers so long as you use a surround sound processor to decode the required four signals from the two channels in the recording, but the trend in the industry today is to assemble systems out of specialized components—particularly in the case of the speakers. Very specific performance characteristics are required to optimally reproduce a movie soundtrack, and not all speakers or amps are suitable. Remember, home theater amps and speakers are usually required to do double duty as music reproducers. All Harman Kardon home theater products offer outstanding music performance and outstanding theater sound reproduction.

Harman Kardon AVR 25 Audio Video Receiver

As a rule a home theater system should use speakers designed for this application, and optimized by the manufacturer to work together as a system. Mixing different home theater speakers rarely gives good results since speaker to speaker discontinuities tend to compromise imaging.

## Channels Out of Two

The third element in creating a successful home theater system is the calibration and tuning of the system in the room. This is done through the A/V receiver or surround sound decoder. In order to understand the calibration process, it helps to understand how a decoder operates. It also helps to realize that not all decoders are equivalent.




Fantasia *was the first film shown publicly with stereo sound, utilizing three optical tracks on separate 35 mm film played in sync with picture.*

FIG. 8d

First, let us examine decoders in general.

If you remember the quadraphonic sound era of the seventies, you may recall that many multi-channel sound software formats were offered to the public (Fig. 8d). Each required a different decoder. However, virtually all surround decoders sold today are designed to work with source material encoded through the Dolby type matrix surround process where left, center, right, and surround channels are encoded into stereo left and right. Most major release feature films made since 1978 include surround encoding, and that encoding follows through to the broadcast video release of these films. In addition, a growing number of television series and specials are now produced with surround encoding. A good source for a list of these programs is Dolby Laboratories (415-558-0200).

All current four channel matrix decoders derive the additional channels from the stereo pair by combining and recombining the left and right channels in various phase relationships. In the Dolby matrix (the only one with which we'll be concerned), the basic formula for getting the center channel is simply to sum left and right in phase into a mono signal. The surround is derived by combining left and right out of phase so that the common information in both channels is canceled out, and only a difference signal remains. This simple "left plus right, left minus right" matrix forms the basis for Dolby decoding of two channels into four. When no other processing is used, the resulting decoder is known as a passive decoder.

Passive decoders were once common in the consumer market, but have almost disappeared from the market because the simple matrix by itself only yields 3dB of channel separation between adjacent channels i.e. between left and center, center and right, and right or left and surround. Such limited separation simply isn't enough to reproduce precise pans or to place individual sounds in specific locations in relation to onscreen action. Something more is needed. That something more is high separation decoding which today generally means Dolby Pro Logic™.

High separation decoding grew out of the old quadraphonic sound phenomenon, and was simultaneously developed in the 1970s by Sansui Corporation in Japan and by Peter Scheiber, an American mathematician and inventor, in the United States. Scheiber holds the key patents in this area, and the high separation decoders in use today flow out of his work. The first high separation decoders were intended to work on the old QS and SQ quadraphonic music formats, but basic technology was soon adopted by Dolby for motion picture use.

High separation decoding was introduced to the consumer market—initially for the old SQ format—by two American companies, Audionics of Oregon and Fosgate Research, which merged in the eighties. Subsequently Fosgate•Audionics pioneered high separation decoding for home use and has consistently led the industry in surround technology. Fosgate•Audionics is now part of the Harman Kardon family and we are able to draw upon its accumulated experience and unmatched expertise in this area.

**Dolby Pro Logic™**

We have already mentioned the Dolby Stereo optical film recording system. This system was brought to its present form in 1976, and in the years since has been used on over five thousand motion picture releases. This theatrical Dolby recording system always used a high separation decoder, one that is closely related in principle of operation to the Fosgate•Audionics' consumer products.

During the early eighties Fosgate•Audionics and a number of other companies introduced high separation decoders specifically designed to decode Dolby Surround video soundtracks. Eventually, this led to the start of the home theater market. With the growth of home theater, Dolby Labs decided to offer its own expertise in decoder design to the consumer electronics industry, by providing detailed design criteria to manufacturers in return for a licensing fee.

A decoder that meets Dolby Pro Logic licensing requirements offers performance that is quite close to that of a Dolby theater decoder. It is free from the noise problems and jerky pans that plagued early high separation decoders. Pro Logic has also set a high minimum standard for the industry and has helped to make high performance decoding affordable for the average consumer. Thus, it has played an extremely important role in the growth of home theater in this decade.

Every Harman Kardon surround processor, A/V receiver and A/V integrated amp displays the Dolby Pro Logic logo, and meets or exceeds the Dolby standard for performance. The addition of Harman Kardon fully discrete, low feedback amplifier sections delivers an even higher level of performance.

**How High Separation Decoding Works**

A full explanation of the operation of a high separation decoder, also called a logic steering decoder or matrix multiplier, would require a lengthy technical paper (Fig. 8e). However, the basic principles involved are reasonably simple.

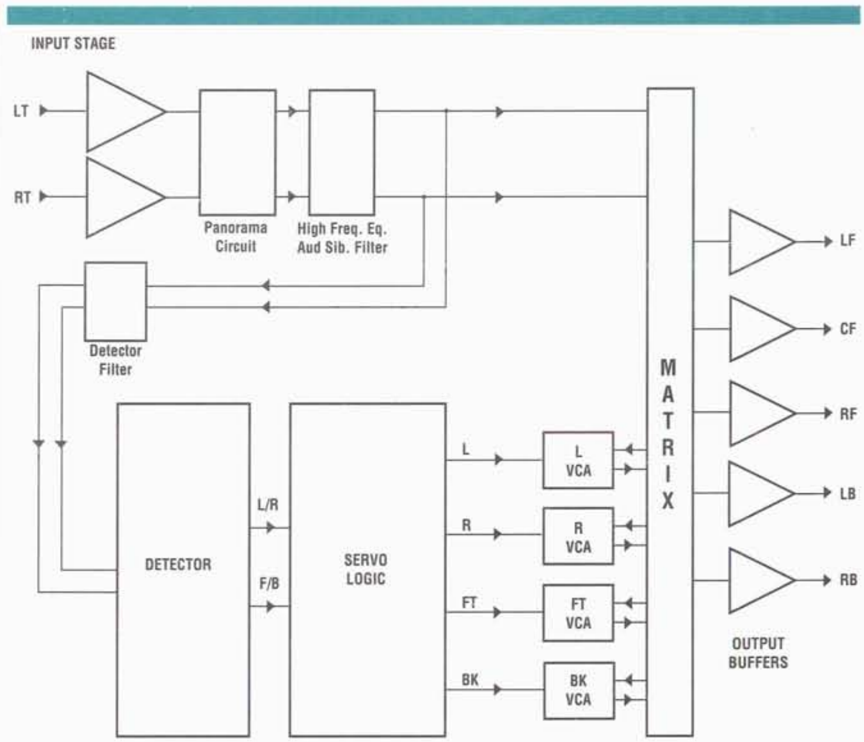



FIG. 8e

© 1992, Pat. Nos. 4,932,059, 5,172,415, 5,263,087, 5,307,415

A high separation decoder routes the four channel output of a simple matrix decoder, which, incidentally, is always part of the first stage of the high separation decoder, to two additional types of signal processing. These are known respectively as vector cancellation and gain riding. Together these two processes create the effect of four highly separated channels.

As we have seen, adjacent channels derived by the passive matrix have only 3dB of separation between them. However, between alternate channels, either left and right, or center and surround, the separation is nearly equal to the basic stereo separation of the recording medium, which is almost infinite in the case of a laserdisc's digital soundtrack, and better than 40dB for VHS Hi-fi. Thus, to get four highly separated channels from a Dolby encoded movie soundtrack, one need only work on the two pairs of adjacent channels since the alternate channels are already highly separated.

In actual fact, most decoders only operate on one pair of adjacent channels at a time, which is generally all that is necessary. This is because a movie sound recording mixer will normally not try to pan between two adjacent pairs of channels simultaneously. Instead the mixer will pan only one dominant sound—often a sound in motion.

If the mixer is panning a sound moving across the screen—left to right—the trick is to achieve high separation first between left and surround and left and center, and then, after the left channel has fallen relative to center, between center and left and center and right. Then, as the right channel rises in output, separation must be maintained between right and center and right and surround.

Conversely, in the case of a pan in depth from center to surround, high separation must first be obtained between center and left and center and right, and then as the balance shifts toward the rear, between surround and left and surround and right. The channel whose output is dominant must always be flanked by two highly separated adjacent channels.

How is all of this accomplished?

We have already mentioned vector cancellation and gain riding, but before we explain these how they are applied to the outputs of the passive matrix to produce high separation, we must first examine a process known as dominant channel detection. (Remember, the dominant channel is the one that is separated out during high separation decoding.)

This process is just what the name implies, and it takes place in a detector circuit that compares the relative levels of all four channels. It then selects the channel whose output is greatest at any one instant. (Even with the low separation passive matrix, channel dominance is evident to the detector circuit.) Once the decision has been made as to channel dominance, the matrix vector cancellation circuits can go to work.

To see how these processes actually work, let's suppose that the left channel is dominant at one particular point in time, but the process is exactly the same for any channel that happens to be dominant.

In this instance the two adjacent channels are surround and center, both of which are heavily contaminated with left channel information known as crosstalk. In addition, left channel signal itself contains a large amount of center and surround channel crosstalk. The vector cancellation circuit removes a large amount of this crosstalk from all three channels—from the dominant left, as well as from the adjacent surround and center. This is accomplished by feeding phase inverted left channel information to the surround and the center, while simultaneously feeding phase inverted center and surround channel information to the left channel. This approach cancels out information the three channels have in common leaving us with only what is distinct to each of the three.

Unfortunately, the cancellation signals themselves represent the original low separation outputs of the passive matrix, so they produce an unwanted side effect: A reduction in the level of the channel they are acting upon. For example when inverted surround information is mixed with the left channel, it cancels out much of the surround channel crosstalk contaminating the left channel. However, it also cancels out some of the pure left information because the cancellation signal itself contains left channel crosstalk. That's where gain riding comes in.

The gain riding circuit remembers the relative levels of the outputs of the passive matrix, and it selectively boosts the outputs of the vector cancellation section to restore all four signals to their original levels. If this were not done, the increase in separation would be matched by a drop in signal level in the dominant channel. If that happened, localization of sounds to the dominant channel would not be much better than in a passive decoder. Gain riding is what makes the process of vector cancellation work properly.

Before we leave the subject of high separation decoding, mention should be made of one further aspect of these systems, the surround channel delay line. This circuit delays the signal to the surround speakers by a few thousandths of a second. The purpose of this circuit is to recreate in a home setting the delayed arrival of surround information in a movie theater. This delay is a natural result of the distance between the surround speakers from the theater's seats. The normal Dolby setting for the rear channel delay is 30 milliseconds in a home system, but the decoders in most products allow for some adjustment in the delay time to accommodate varying room dimensions and listener tastes.

Some room layouts do not permit the use of rear channel speakers, and some consumers might not be able to afford rear speakers at the time they purchase their first product with a surround decoder. To accommodate those customers, some entry level products include the "Dolby 3 Stereo" mode, which adds a float channel but no surround channels.

A home theater set-up with only the three front channels is not going to provide the full movie theater experience, but it will still be capable of handling lateral pans correctly by providing a kind of sound stage that goes beyond simple two channel stereo. It will also provide a more stable center image than a two speaker arrangement. It will also offer increased dialog intelligibility through the use of the center channel.

For a media room where rear channel speakers are just not possible, the Dolby 3 Stereo mode is a good option to suggest Of course, it's also cost effective.

**A Complex and Difficult Technology**

Once you understand the basic operating principles of high separation decoding, the process seems rather straightforward, but making it work flawlessly has been a major challenge for engineers and designers. Early high separation decoders added unwanted noise to the signals when the logic circuit was working. They also suffered from poor performance when the recording was less than perfect. In recent years, however, the performance of surround decoding circuits has been dramatically improved.

The improvement in the average level of decoder performance has been due to the development of a consumer version of Dolby Labs' high separation theater decoder circuitry, which forms the basis of Dolby Pro Logic. Most decoders on the market today meet Dolby licensing requirements and thus perform quite well.

Although Dolby Pro Logic is as an industry standard, it does not define the ultimate level of performance available in a decoder. The ultimate is defined by the products of two companies, Fosgate•Audionics and Lexicon, both of which are now part of the Harman International family.

These two companies manufacture surround processors that provide superior separation with both low level and low frequency signals, cleaner less colored sound reproduction, greater ability to track highly dynamic, rapidly changing signals, and more accurate reproduction of poor soundtracks. When cost is no object, they are indisputably the best. When cost is a concern, which is certainly the case with most sales, the much more affordable Harman Kardon A/V products represent the logical choice. Again, as in so many other Harman Kardon components, we have taken the engineering techniques used in the very best esoteric audio products and adapted them to popularly priced, mass produced items. Nowhere does this approach provide greater value than in home theater, where systems require more components and where cost effectiveness can save the customer literally many dollars.


Harman Kardon AVP1a Surround Processor

## OTHER FUNCTIONS OF DECODERS IN HOME THEATER SYSTEMS

The surround sound decoder is truly the hub of a home theater system. Most of the electronic components in the system are connected to the decoder so that the system as whole can be controlled through it and made to perform in a variety of different ways.

### System Calibration

An unexpected benefit of surround sound decoders is the ease with which a complex system can be balanced and calibrated with them. All Pro Logic decoders include a test signal generator that produces a pink noise signal which appears successively in each channel. Since the level of this noise never varies at the output of the decoder, it can be used as a reliable reference for balancing a system made up of amplifiers of different gain factors, or even of different loudspeakers. The individual output level or volume controls for each decoder channel are simply adjusted until all speakers emit the noise signal at the same level, and the system is balanced.

It is important to explain this feature to your customers, as the small amount of time they spend setting the levels in their system at home will yield them substantial dividends in listening enjoyment. Of course, any complex system installed in a home system should be calibrated with a sound pressure level meter. A popular tool for this purpose is widely available for under $50.

Beyond that, decoders offer a variety of playback modes, each of which alters the sound of the system. These modes are worth discussing in general insofar as they have led to high degree of bewilderment among consumers confronted with them.

### Surround Sound Playback Modes

Any Dolby Pro Logic decoder is obviously capable of decoding encoded movie sound-tracks; after all, that is the decoder's primary purpose. But many home theater products also have various music modes with names such as stadium, church, hall, rock, classical, etc. There is no standard nomencla-ture for these modes or a consistent industry approach to their operation, but we can describe in general terms what is being attempted.

Although most movies, many TV programs and a few hundred music CDs have surround encoding, the vast majority of music CDs, as well as virtually every vinyl record and audio cassette produced, do not. Without this encoding, it is difficult for the decoder to function properly. Playing an unencoded source in the Prologic mode will rarely produce a good result. To compensate for this, most A/V or home theater products include these additional modes for use with unencoded stereo material.

In general, these music modes are attempts to simulate the ambience of various listening spaces where live performances would occur. Normally such spaces are considerably larger than a home entertainment room, and thus the sound reflected from the walls takes a relatively long time to reach the listener and a relatively long time to fade away. In other words, such listening spaces have a lot of reverberation compared to a living room.

This reverberation can be suggested after a fashion by sending a sequence of delayed signals through the surround speakers in a home theater system, and when this is done, the surrounds become a type of electronic echo chamber. By varying the number, intensity, and timing of the "echoes," the processor attempts to recreate the sound of a small club, a large church, a concert hall, and so on.

In most A/V products, particularly A/V receivers, these additional modes are created through the use of Digital Signal Processing, which is frequently referred to as "DSP." What the DSP does is alter the audio signals in such a way that the resulting sound best imitates the type of room selected. Most often these modes are used with music sources that do not contain the surround encoding needed for proper multi-channel reproduction. Thus, they do provide a means for customers to use their surround systems with program material other than movies or TV shows.

Some audiophile purists are vehemently opposed to the use of artificial surround processing, as it is easy to overuse the delay and create very unnatural sound. However, many consumers like the effect, as it does seem to create a sound field that totally envelopes the room. You should be aware of the impact of the various modes offered on different A/V products, and audition them before you explain them to customers.

You will note that some modes do sound better than others. The circuit design in Harman Kardon A/V products is specially tailored to avoid the unnatural effect of DSP, while delivering the features that customers insist on.

**Fosgate•Audionics**

**Surround Modes**

The special surround modes offered on Fosgate•Audionics products, and selected Harman Kardon models differ greatly from conventional DSP driven modes. They extract the ambience information that is present in almost all recordings and process it through exclusive, patented circuitry. The result is a careful steering of certain parts of a program to the center and rear speakers, something that is almost impossible to do with conventional, off-the-shelf DSP chips.

These surround modes do work well with almost any type of program source, whether it carries the surround encoding matrix or not. They are a very special feature that is exclusive to Harman International products.

## SUMMARY

Home theater represents a special challenge to the retail community. On one hand, the components of a home theater system represent the greatest sales opportunity since the introduction of the VCR or compact disc. On the other hand, it involves components and concepts that have the potential to cause a great deal of confusion among consumers.

Because it is an evolving field, you will not only have to become familiar with the information presented here, but you will have to keep up to date on high definition television, multi-channel theater sound, digital audio systems and a host of other emerging technologies. The media will keep your customers informed; you can do no less for yourself.

However, as an expert you will have an unmatched opportunity to serve your customer by tailoring the products you sell to their needs and budget. Harman Kardon products are unique in the value that they deliver to the home theater customer. Your ability to properly explain their features, benefits and technology to a prospect will greatly improve your chances of making a sale, and creating positive word of mouth referrals from satisfied customers.

# A2

APPENDIX

## COMPACT DISCS IN DOLBY SURROUND

Depending on the configuration of your store's sound room, it may not always be convenient to demonstrate a surround sound product where there is a VCR or laser disc player available. Fortunately, that is no longer a problem!

A growing number of CDs from a variety of labels are now being recorded In Dolby Surround. This means that you can use any CD player in conjunction with a Harman Kardon A/V product to show your prospects the benefits of surround for *all types of music.*

Letting a prospect hear their favorite music, be it jazz, classical, rock or show tunes in full surround will help you make the case that surround products are for more than movies and TV shows.

This list is only the tip of the iceberg, as more surround encoded discs are being released all the time. Look for the "Dolby Surround" logo on a CD's artwork. That is your clue that the disc should be added to your demo list. This list is also being constantly updated by Dolby Laboratories. Updates on program material in all formats: VCR, Laser Disc, CD, Video Game and TV may be obtained from the Dolby Laboratories forum on the America On-Line data service. Use the keyword "Dolby" to access the listings.

The following compact discs are recorded in Dolby Surround for multi-dimensional playback through your Dolby Surround decoder:

**CONCORD JAZZ**

Frank Wess
Entrenous
CCD-4456

Mel Torme
Night at the Concord Pavilion
CCD-4437

Mel Torme/George Shearing
"Do" World War II
CCD-4471

Poncho Sanchez
A Night at Kimballs East
CCD-4472

Various Artists
Live at the 1990 Concord Jazz Festival — Volume 1
CCD-4451

Various Artists
Live at the 1990 Concord Jazz Festival — Volume 2
CCD-4452

Various Artists
Live at the 1990 Concord Jazz Festival — Volume 3
CCD-4454

**PRO ARTE**

Columbus S.O./Badea

Bartok: Miraculous Mandarin, etc.

CDD 535

Copenhagen Phil./Entremont

St. Saens: Organ Symphony

CDD 534

Dallas S.O./Mata

Spanish Orchestral Music

CDD 536

Dallas S.O./Mata

Stravinsky: Petrouchka

CDD 537

Peter Nero and the

Columbus Symphony

Anything But Lonely

CDD 522

Helsinki Philharmonic/Comissiona

Classical Storm Music

CDD 530

Rochester Phil./Elder

Gilbert & Sullivan Overtures

CDD 533

San Diego S.O./Schifrin

Those Fabulous Hollywood Marches

CDD 504

San Diego S.O./Talmi

Gliere: Symphony No. 3

CDD 538

Various American Orchestras

A Broadway Spectacular

CDD 529

Various American Orchestras

Orchestra Spectaculars of

Classical Music

CDD 528

Various Orchestras

The American Home Video

Classical Album

CDD 529

Vienna C.O./Entremont

Mozart: Symphonies 40 & 41

CDD 531

Vienna C.O./Entremont

Mozart: Eine Kleine Nachtmusik, etc.

CDD 532

**RCA VICTOR**

Arthur Fiedler, The Boston Pops

Motion Picture Classics, Volume Two

60393-2/4-RG

Arthur Fiedler, The Boston Pops

Motion Picture Classics, Volume One

60392-2/4-RG

Charles Gerhardt,

The National Philharmonic

Now Voyager, The Classic Film

Scores of Max Steiner

0136-2/4-RG

Charles Gerhardt,
The National Philharmonic
Captain From Castile, The Classic Film Scores of Alfred Newman
0184-2/4-RG

Charles Gerhardt,
The National Philharmonic
Casablanca, Classic Film Scores for Humphrey Bogart
0422-2/4-RG

Charles Gerhardt,
The National Philharmonic
Gone With The Wind,
The Classic Film Score by Max Steiner
0452-2/4-RG

Charles Gerhardt,
The National Philharmonic
Citizen Kane, The Classic Film Scores of Bernard Herrmann
0707-2/4-RG

Charles Gerhardt,
The National Philharmonic
Sunset Boulevard, The Classic Film Scores of Franz Waxman
0708-2/4-RG

Charles Gerhardt,
The National Philharmonic
Classic Film Scores for Bette Davis
0813-2/4-RG

Charles Gerhardt,
The National Philharmonic
Elizabeth & Essex, The Classic Film Scores of Erich Wolfgang Korngold
0815-2/4-RG

Charles Gerhardt,
The National Philharmonic
Spellbound, The Classic Film Scores of Miklos Rozsa
0911-2/4-RG

Charles Gerhardt,
The National Philharmonic
Captain Blood, Classic Film Scores for Errol Flynn
0912-2/4-RG

Charles Gerhardt,
The National Philharmonic
Lost Horizon, The Classic Film Scores of Dimitri Tiomkin
1669-2/4-RG

Charles Gerhardt,
The National Philharmonic
The Spectacular World of The Classics Film Scores, Sampler
2792-2/4-RG

Charles Gerhardt,
The National Philharmonic
Close Encounters/Star Wars, John Williams' Classic Film Scores
2698-2/4-RG

Charles Gerhardt,
The National Philharmonic
Star Wars: Return of the Jedi, John Williams' Classic Film Score
60767-2/4-RG

Charles Gerhardt,
The National Philharmonic
The Sea Hawk, The Classic Film Scores of Erich Wolfgang Korngold
60863-2/4-RG

David Raksin,
The New Philharmonia Orchestra
Laura/Forever Amber/
The Bad & The Beautiful
1490-2/4-RG

Henry Mancini,
The Mancini Pops Orchestra
Mancini in Surround —
Mostly Monsters, Murders & Mysteries
60471-2/4-RC

Henry Mancini,
The Mancini Pops Orchestra
Mancini's "Monster" Hits (Collector's Edition, Glows In The Dark)
60577-2/4-RV

Henry Mancini,
The Mancini Pops Orchestra
Cinema Italiano —
Music of Ennio Morricone & Nino Rosa
60706-2/4-RC

Henry Mancini,
The Mancini Pops Orchestra
The Pink Panther & Other Hits —
Newly Remixed Original Recordings
7863-55938-2/4

Isao Tomita
Kosmos
2616-2/4-RG

Isao Tomita
The Tomita Planets (Host)
60518-2/4-RG

Isao Tomita
Pictures at an Exhibition (Mussorgsky)
60576-2/4-RG

Isao Tomita
Firebird (Stravinsky)
60578-2/4-RG

Isao Tomita
Snowflakes Are Dancing (Debussy)
60579-2/4-RG

London Cast Recording
Into The Woods —
Stephen Sondheim/James Lapine
60752-2/4-RC

Original Soundtrack Recording
Altered States, John Corigliano
3983-2/4-RG

Various

Silver Screen Classics — 4 Classic

Film Scores (Collector's Edition)

60763-2/4-RG

Various Artists

The Home Video Album

60354-2/4/9-RC

# A3

APPENDIX

## THE LUCASFILM THX PROGRAM

Harman Kardon, and its sister companies JBL, Fosgate•Audionics, and Lexicon are licensees of the Home THX program. This program, which sets detailed standards for home theater performance and certifies equipment that has met these standards, has achieved great name recognition among the public at large. It has received enthusiastic response from reviewers and discriminating consumers alike. THX certification is becoming increasingly important for products competing in the high end and upper middle portions of the home theater marketplace.

First a little background:

THX began as a program for movie theater sound systems only. George Lucas, producer/director of "Star Wars" fame, felt that soundtracks for the movies he produced were not being played back with the intended impact and clarity in most of the movie theaters, so his company instituted a comprehensive program to certify theater sound systems. The program involved laying out acoustical requirements for theaters, licensing equipment, then setting installation standards.

Theaters that gained THX certification generally found that they attracted more customers, and movie goers learned to associate the famous THX trailer with a more exciting film experience. Today over 700 theaters worldwide are THX certified.

The growth of the home theater phenomenon in the late eighties and early nineties prompted Lucasfilm to offer a home version of the program. Like the professional standard, Home THX entails the licensing of individual components, but it does not involve system certification as such. A consumer can have a complete THX system, or purchase only selected THX approved components.

THX licensing of home audio components currently covers decoders, amplifiers, loudspeakers, audio cables, projection screens and laser disc players.

Overall, THX strives to replicate the sound created on a dubbing stage, in a theater or in a home (Fig. A3a). To accomplish that, Home THX systems must exhibit wide dynamic range, especially at low frequencies, as well as smooth frequency response, controlled loudspeaker directivity, and a very specific equalization curve.

**Wide Dynamic Range**

The requirement for wide dynamic range is fairly obvious. Modern movie soundtracks tend to emphasize dramatic sound effects such as crashes and explosions, and the master soundtrack recordings are encoded in either Dolby SR or digitally recorded to capture the full

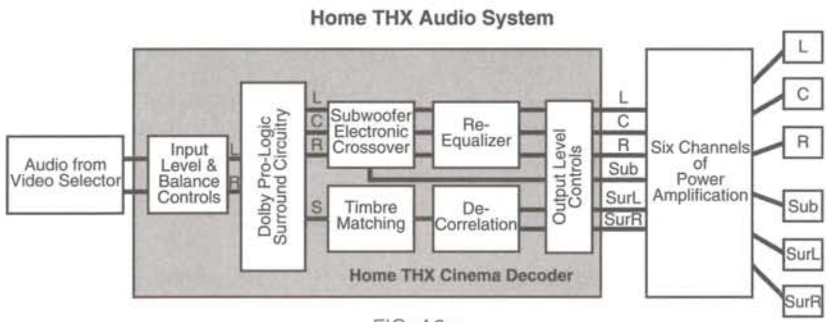



FIG. A3a
In the Home THX Audio System, left and right stereo signals are decoded to produce left, center, right and surround signals. Home THX then processes these signals to produce left, center, right, subwoofer and a pair of decorrelated surround channels.

dynamic range of these effects. VHS Hi-fi and laserdisc are both capable of reproducing the full dynamic range of most movie soundtracks. It follows that for accurate reproduction, the speakers must have similar dynamic range, and the amplifiers driving them must be capable of supplying the power the speakers require to reach full output. Since much of the sound energy in modern movie soundtracks is concentrated in the low frequencies, a speaker must produce high, undistorted sound pressure levels at those frequencies, in addition to the midrange where most speakers are normally tested.

**Smooth Frequency Response**

The requirement for smooth, level frequency response is not much different from what one would expect from a high fidelity music speaker. In the case of THX certified speakers, the smoothness of response must extend to listening positions that are far off axis, since the sound system is likely to be used to entertain a number of people at once.

**Controlled Directivity**

Controlled directivity refers to the manner in which the loudspeakers disperse sound into the listening space. Here THX certified equipment tends to differ a bit from ordinary speakers in a number of respects.

Requirements are quite different for front speakers and surrounds.

In order to reproduce movie quality sound for a number of listeners, the front speakers, very generally, must achieve wide coverage in the horizontal plane, but should be restricted in their off axis output within the vertical plane. A speaker with broad vertical dispersion will bounce a lot of sound off the floor and ceiling and these reflected sounds tend to degrade spatial imaging.

Surround speakers, on the other hand, should scatter sound as widely as possible. Ideally all of the sound reaching the listeners should be indirect, reflected sound. This kind of dispersion pattern is best achieved by designing the speakers to be dipolar. That is, they produce sound from both the front and back. The speakers are then placed to the sides of the listening positions so that the main outputs of the speakers strike the back, side, and front walls of the room before they arrive at the listening positions. This ensures that surround sounds appear nondirectional and diffuse, seeming to come from all around. And to further facilitate such a dispersion pattern, THX mandates special signal processing that "decorrelates" the signals to the surround speakers so that the speakers themselves can't be identified as sound sources.

**Equalization**

The THX equalization curve is intended to adjust the tonal balance of video soundtracks to approximate in the home what you would hear in a properly equalized theater.

Movie theaters are designed and tuned to accommodate the requirements of large room acoustics. Those practices and the nature of the systems used to build and calibrate movie theater sound systems result in soundtracks that may tend to be a little on the bright side if played back through a standard home audio system. Normally such soundtracks are not re-equalized for video. When they are played back over a flat speaker system they often sound too bright in a home setting where the smaller room volume will have little effect on the highs. THX mandates a gradual reduction in high frequency output to make up for this.

**Decoder Requirements**

Home THX processors must include several functional requirements and they must also meet specific performance criteria. Of course, they must include Dolby Pro-Logic decoding. In addition they are required to have circuits that include front change re-equalization as well as surround timbre match and decorrelation. A high performance subwoofer crossover must also be provided.

**Amplifiers**

THX amplifiers must have abundant power, low distortion, flat frequency response, and must be stable under a wide variety of real world speaker loads. These are precisely the performance characteristics that Harman Kardon has been building into their amplifiers for over forty years. Refer back to Chapters 3 and 4 of this book.

**Speakers**

THX loudspeaker requirements naturally vary according to whether the speakers will be used for the three front channels, the subwoofer channel, or the surround channel. Front speakers must have identical characteristics for all three channels, and must exhibit the directivity patterns described above.

They must also be capable of low distortion at high output, and must present an easy to drive electrical impedance to the amplifier. Subwoofers must have output down to 20Hz, and must also be capable of high output at low distortion. Surrounds, in addition to being dipolar, must be timbre matched to the front speakers, that is, they must be designed to produce the same frequency balance as the front speakers when placed behind the listener.

**A Standard, Not an Upper Limit**

THX, like Dolby Pro Logic, constitutes a minimum performance standard not a complete specification, and certainly not all THX systems are equivalent. The THX approved products offered by JBL, Harman Kardon and Fosgate•Audionics offer a level of performance in the home which rivals that of virtually any commercial movie theater.

At Harman Kardon we feel that the THX standards are generally in harmony with the audio design philosophies we've been advocating for over 40 years. At the same time there are aspects of Harman Kardon component design that go beyond the basic THX standard and make for an even higher level of performance—and here we refer to the design philosophies covered in Chapter 2. We're very proud that some of our products bear the THX logo (Figure A3b), and we're just as proud of our own unique achievements in design. And we strongly believe that at any given price point Harman Kardon THX certified products represent the best value offered by anyone in the industry.

FIG. A3b