WHAT ABOUT LOUDSPEAKERS?

by Irving "Bud" M. Fried

What can be said about loudspeakers that is new and interesting, different from what has been said before? Every serious researcher and writer on the art continually faces that question. As do marketers, be they advertising experts or salesmen on the retail floor. Indeed, an entire industry's fate rests upon an answer to that question; for, if there is really nothing new and interesting to say, there is really no progress in the art, no reason for the professional or the music lover to listen or read. He can presume that his current loudspeakers have already reached the limits of the art of music reproduction.

The author has asked himself that question for some forty years, and found new and fascinating answers forty years ago, ten years ago, last year, and this year. The science and the art of the loudspeaker are ever advancing, and so, there is purpose in this article.

Anyone who has been subjected to the process of education, particularly at the college or graduate levels, knows that the asking of the proper question is part, an important part, of finding the answer. One of the difficulties in discussions of the loudspeaker, in the author's opinion, is that the proper questions are not being posed, for one reason or another, and so results in today's loudspeakers not being as good as they could or should be. Stated somewhat differently, many of today's loudspeakers are not being designed by well-educated people, by people who know the history, the technology of the past, today's knowledge, and just how a loudspeaker should be superior to those of the past.

On the presumption that many or most of you are well educated, somewhat skeptical, but interested, and responsive to facts; the author proposes to ask questions and answer them in an "educated" way, in terms of what is known and accepted among the "serious" followers and designers of the art of the loudspeaker, those in various countries who are actually researching the sciences of acoustics, and psychoacoustics. While not every answer to be offered is unanimously accepted everywhere, no answer will be given if it is not in fact accepted by the preponderance of informed, or as we say, "educated" researchers; unless that answer is specifically termed "still-controversial" or, "only in the author's opinion", you can rest assured that responsible people everywhere will agree in whole or in the major part. It has been and is forever interesting to the author, that when this select group of "educated" researchers communicate with each other, whether they be oriented toward electrostatics, or dynamic speakers; or English, American, or Scandinavian, or Japanese; there is very little disagreement or drawing of battle lines. Rather, there is general agreement on what is important, what the problems are that remain unsolved. The chaotic disagreements that many writers in the popular journals evidence-the statements, it is all what you like, it is personal, subjective-seem nonexistent among the real pursuers of the truth.

Well, the reason for that, the difference between what is known and what is supposed to be known, is the familiar (to sociologists) concept of "cultural lag", which occurs in all disciplines, being the time lag between research knowledge and common knowledge. Then there is that grievous problem of commerce, to "sell what you got", or what they ask for", etc.

We ask our first question: "What is a loudspeaker?"

Right away, we understand why there seems to be so much disagreement at the commonly accepted (as opposed to "educated") level. It is because, in the layman's press, there is so much confusion concerning what a loudspeaker is! The author has spent years and years "Reviewing the Reviewers", and concludes that the answers to our question fall into essentially three separate categories, as follows:

1) A loudspeaker is an electro acoustic transducer, which converts an electrical input into a near exact acoustic analog, then feeds into a room and to the auditioner with as little change as is possible at that stage of the art; so that, from the proper input, the perception by the listener will be similar to that from a live program.

2) An electro acoustic transducer which produces an altered acoustic analog to the auditioner; whether done purposely, from ignorance, or from pure circumstance; which is then attractive to a certain auditioner, conditioned by time, place, education, neurosis, sexual disposition, or general life style.

3) A pure entertainment medium, with no reference to live or original musical values. In this category are all loudspeakers which apparently are to titillate with synthetic sounds, as in a disco; or to provide an altered state of reality, as does the drug scene.

We might write about all of these. Indeed, the author has written about them in the past. However, in the interests of the reader, and hoping not to add to the general confusion, the author intends to write only about the first category, sometimes called the "monitor" loudspeaker. A further, selfish reason for writing about the "monitor", or "accurate" loudspeaker, as we will call it henceforth, is that the author has written about them before, several times; so that he can draw on his own past writing and the references; and thus he does know more about them, from a very long term involvement in them.

We commence with a firm resolve, that we will try to avoid old chestnuts; indeed, if we don't, we expect that many of you will stop reading on to the bitter end. We will stay with factual data, much of it from professional journals here and in other countries; some of it in the heads of various people who have communicated it to the author, prior to its publication (a fast moving art, this!). We will try to inject a bit of humor into the incongruities we will encounter-if you don't have a sense of humor, our friends tell us, you can go crazy with frustration; vis a vis what is possible and what they must manufacture "for the market". So, much of what we will cover is not particularly commercial-it won't "play in Peoria", to repeat a famous phrase from a marketer in another field. Much of what you will learn, we hope, is really laboratory data, which is several years off from being made into a market loudspeaker.

As we begin, we must mention the groups that have done work in the past, and continuing in some cases into the present, on pure research into loudspeakers and sound, with no direct relationship to any commercial product. We mention the early work of

Western Electric, in the 1930s, into record reproduction, sound, and "stereo" (the latter in the incredible Stokowski concerts in Washington and Philadelphia, in 1931); and we must pay special homage to the work of the BBC and their researches into monitoring loudspeakers. We also must, in passing, note the work of Bruel & Kjaer, the test instrument makers; and the other pioneering work in Denmark, by the relatively unknown Madsen, Amneus, and Lian. We will return again and again to some of these names, so keep them in a special section of your memory bank.

Right now, we must commend the BBC, which developed the modem concept of the monitoring loudspeaker, the one we have defined above. They did it by comparing live performers and alternate loudspeaker designs in an adjacent room (very few speaker designers have the facilities of the BBC, and the research money!), and by asking the right questions. How do we know they asked the right questions? We know because they asked the questions the Author asks-they agree with the author! Instead of asking, "Which sound do you like?" (a common question in hi-fi showrooms), they asked, "Which sound is closest to the performers in the other room?" So, we see again the virtues of being able to ask the right questions.

To answer those questions, they did use informed, "educated" (accustomed to music reproduction and live music) people. The author must contrast this situation with one of which he learned many years ago, re the design of a famous "air suspension" loudspeaker; where the designer took office girls and secretaries and offered alternate designs, accepting the one they liked! The BBC pointedly ignored the crassly commercial, "Will it sell?, "Bottom Line", "What will this or that reviewer say?", etc.

So, we now ask the right question, and we have people to whom the answer is important-you, who are not swayed, we trust, by lack of education and preference for the banal; but who are searching for more accurate reproduction. Before passing on, the author is always delighted to find that even the employees of "bottom line" companies, the engineers, agree on ends and means. He will never forget the lecture he delivered to the Boston Audio Society, in the bastion of the Lowells, codfish, and "air suspension"; when, after a comparative demonstration, the employees of the foremost "air suspension" companies voted overwhelmingly for the "superior realism" of an early transmission line system!

Still, before we begin, we will again subdivide our answers to question thusly: a) The accuracy of the transducing system, its ability to reproduce acoustic wave forms and wave fronts; and, b) The interaction between the transducer, the room, and the auditioner; how we can control this; what laws of acoustics and perception apply, which we can use so that the auditioner believes he is approaching live performance-our goal.

It is the author's fervent belief that designers of loudspeakers just don’t do their homework on this latter subject, how the human hearing mechanics are able to perceive sensations of sound, and judge them. As an example, the hearing mechanism senses two laterally spaced sources of the same data as "unnatural", if they are close to each other (if they are far from each other, it becomes "stereo"). Yet, many loudspeakers have two closely spaced lateral sources of the same data-all wrong!

We now begin our treatment of the "accurate" loudspeaker, as defined. What detracts from accuracy? By definition, "distortion" is a deviation. So, we must look at various kinds of distortion, methods of measuring it; and then we must weigh these kinds of distortion, in light of prevailing knowledge; because not all kinds of measured distortion have the same deleterious effect on the hearing process, on the perceived quality of reproduction.

A recent article treated measured low frequency harmonic distortion as a significant mark of merit in loudspeaker design. Yet, the preponderance of researchers known to the author give low frequency harmonic distortion only short shrift! The preponderance of informed opinion is that high level, low frequency harmonic distortion is interesting, but not significant, unless further investigation correlates it with a basic non-linearity in the system. It can be an indication of limited power handling, which is unrealistic, given the size and price of the loudspeaker; it can merely indicate that the system is not "flat" at that frequency, so that the harmonics are reproduced with more relative energy than the fundamental-which may not be a defect at all; indeed, later, we will show that such a lack of flatness may indicate superior accuracy for that category of loudspeaker!

Everyone agrees that measured harmonic distortions are much more significant in the midband, where the hearing mechanism is much more sensitive to all forms of distortion. Virtually everyone agrees that measured harmonic distortions in a loudspeaker may be many times that of an amplifier, without detracting from the listening experience. Indeed, one researcher at London-Decca, John Walton, long ago "proved" that distortions approaching 20% harmonic, on a transient, were virtually inaudible. We might question the 20% today; however, distortions on sine waves in anechoic chambers are one thing; distortions on transient data are another, particularly if we listen in a room. Indeed, some of the English researchers, among them the BBC, and the people at KEF, seem to discount low frequency harmonic distortion under room conditions, insofar as the distortion is below 50HZ.

Some English researchers, working with audibility of measured distortions in amplifiers, found that most "golden ear" auditioners actually preferred a listening condition where harmonics were artificially injected into the higher frequencies of music! What does that show us about those who talk of "musical" results - that is precisely how the English "golden ears" categorized the condition of <u>increased</u> harmonic distortion?

Our conclusion must be to disregard harmonic distortion as a vital ingredient, unless it is correlated with non-linearity, which can be measured much more directly, as we shall presently see. We shall also see that in many cases, in order to gain transient accuracy, we must settle for greater harmonic distortion on transients-a very worthwhile tradeoff-gaining something you can hear, and losing something that is really not too audible, on transients.

The BBC long ago discarded sealed enclosures for their studio monitoring systems, because of "non-linearity" at low levels. Harwood of the BBC is adamant on this. In amplifier work, we find a similar trend, in that non-linear distortions at low levels are deemed much more destructive of proper hearing. Ted Jordan, an eminent authority on loudspeakers, a writer of a highly regarded text, talks about several speaker designs (among them, classic air suspension), as having several "conditions" of distortion. Ben Bauer, the late great researcher, inventor, and writer, talked of the "oil canning" effect as being typical of certain low efficiency, heavy cone designs. Ragner Lian of Denmark wrote of "non-linear time delay distortion" in "commercial" loudspeakers (he meant "air suspension"). All of these authorities are really talking about non-linearity, a pernicious effect; all of them are describing classic air suspension-whose advocates persist in calling them "low distortion air suspension" systems! Well!

Frequency response is a more volatile, and interesting subject. Every speaker designer has some equipment to measure frequency response, and every would-be purchaser first wants to know the frequency response, which should be "20HZ to 20KHZ". One authority commented to the author that frequency response is published and commented on, because that is the first test equipment that designers ever get-and, for most of them, the last.

The prevailing popular doctrine is that "flat" response is desirable, with no one being very sure of what is meant by "flat" response, be it axial (on axis) or power (total radiated power at a given frequency). The weight of informed opinion is that the popular doctrine is almost entirely wrong! So, the speaker manufacturer who dares to publish a curve that is not "flat", and who dares to present the speaker which isn't "flat", may be doing a public service. Let us go over the deviations from "flat" which some people would insist are "distortion" (see above), again seeking the preponderance of informed opinion.

Most authorities agree that the very first impression one receives of a loudspeaker is related to its power response. If a loudspeaker has abrupt changes in its power response, because of driver directivity, crossover errors, or enclosure peculiarities, the auditioner immediately notes and rejects-as compared to a loudspeaker that doesn't have the abrupt changes. Thusly, we can understand the seeming popularity among first time auditioners of various reflector and "omni" speakers--on first audition, they are very attractive to beginners. Cunning, crafty marketers who aim at this market have not become poor--or, as H. L. Mencken said, "No one ever went broke underestimating the intelligence of the American people".

However, after one listens for a while, and gains experience at listening, other things about loudspeakers become important and significant-including our present subject, frequency response. So, let us now cover the important further factors pertaining to frequency response.

A few authorities claim that octave-to-octave balance is the prime concern; that each extension upward should be matched with an equal extension downward. Others talk of "natural" program balance, which may and does result in planned deviations from "flat" response.

Let us return again to those fabled BBC researchers. Most of you know the LS3/5A, one of the legends, designed as a mini monitor for the BBC's portable recording vans. It has extended treble response, balanced by a gradual rise centering at 130HZ, for balance. It also contains another BBC technique, the famous "BBC dip". Harwood, a BBC researcher, writes that, in close up monitoring (as described), "flat" midband response makes the instruments of the orchestra appear too "close" and "unnatural"; that a more pleasing overall effect is obtained when a slight trough, no more than a few db., is placed in the one to three KHZ region.

The BBC also concluded that speaker cabinets should be thin walled, rather than thick walled, so that they would resonate at low frequencies, below 1OOHZ, rather than in the midband, and then tuned them for that region.

Thus was created "English sound", all BBC derived, and famous all over the world for its "natural" program balance! It ruled the world for many years, much as did England in an earlier day; like England, it is now going down. But it did well for a long time, as a reaction to the "presence" loudspeakers that had earlier dominated the U.S., the "movie" loudspeakers.

It also made a lot of sense when the drive systems available for the conscientious designer were inferior. A well-known design technique is to depress a region of "trouble" in a loudspeaker, so that the hearing mechanism is not drawn to it. We now know how sensitive the human is to midrange problems; a way of improving the performance in the midband is to insert a shallow dip. Interestingly, the human does not hear shallow dips as such in a loudspeaker-until it is offered a direct comparison, a flat blemish free response in that region-which it first interprets as too much "brightness". These again were the BBC's findings; and can be the bane of modem precision designs, which are matched in retail shops against remnants of the old English tradition (BBC dip), the retailer asking the customer to "choose the one you like"-at first.

As we progress into the modem world, the "natural" balance of the BBC loses its proponents; because it does make for system insensitivity, reduced dynamic range, and poor power handling. Indeed, the latest BBC monitors are a far cry from the earlier ones with the inbuilt frequency distortions we have discussed.

Bruel & Kjaer, the Danish test equipment makers, have researched extensively into those subjects worthy of being measured. On frequency response, their conclusions have been that the loudspeaker total response should slope downward from one KHZ upward, at rates varying from one to three decibels per octave, this apparently a function of the major program material to be reproduced.

Some speaker designers, concerned with the increasing directivity (changes in dispersion) of their speakers as they approach the highs have designed for a rising axial response; either by crossover design or by choice of high frequency units. One very popular dome unit, used extensively in several U.S. made loudspeakers, is sharply peaked at 12KHZ, to "preserve flat power response". The fact that it shrieks because of transient distortion is considered an added plus by some of its users; it helps sell the speaker to innocents (see our discussion above).

We have shown where many deviations from flat response are considered desirable, some of them made by serious researchers; others by manufacturers eager to impress and sell. Can we reconcile the serious deviations, such as from the BBC, B&K.? Yes, we can, historically. As we advance into more refined drivers and new input sources, we will find these august bodies, such as the BBC, changing-but, unfortunately for you, the customer, some of the manufacturers won't, for a long period of time.

Let us retrace a bit, to the first known "flat" loudspeaker that also listened well, for the lessons it may teach us. Some years ago, over a decade ago, the author was developing high quality loudspeakers in England. During the development, he and his co-workers found that a flat axial response was fine, provided that driver behavior was well controlled (all sorts of electrical "damping" circuits were developed for the troublesome drivers of that day), that the crossover networks were refined, that all was "impedance compensated" (we will return to this later), and, most importantly, provided that the loudspeaker could reproduce properly one octave above the limit of the program, and one octave below. To explain, if the program had material from 50HZ to 10KHZ, the speaker had to be substantially flat from 25HZ to 20KHZ.

Such loudspeakers were developed and presented; along with the technical data and research, which was presented in the eminent English journal, "Hi-Fi News". Because of the combined effect, the loudspeakers and the articles, the reviewers and public of that day found them to be more realistic reproducers than anything that was available elsewhere. They became the "reference standard".

As we look backward to those simpler days, for speaker designers, we wonder why we were so successful in influencing the informed opinion of those days-and why we had so little influence on designs from other sources. The answer is, again, history, and how we hear. Remember our statement, that the first thing one notices about a loudspeaker is a function of its "power response", after which may come a more discriminating appraisal of the loudspeaker? The process of evaluation is simply described as proceeding from the general to the particular, from the "mass" of sound to its components, during which we are beginning to be concerned with details such as "Do the drums sound real? Does the triangle tinkle or buzz?"

Well, the author's "flat" loudspeaker did go further; it did reproduce instruments very clearly. The articles on it talked about the "better phase" relations, which we really didn't know much about. However, we postulated that, having wider, flatter band pass than any known loudspeakers, with a comparative absence of "resonant" systems, it must have lower phase distortion in the program range.

So, it then seemed that "phase" has something to do with frequency response preferences. We will go into this subject quite thoroughly later on (this is an incentive to you to keep reading through all this material). At present, we will go over some material concerning methods of getting "flat" response, and their listening qualities. The classic illustration of the text writers is the loudspeaker that is ruler flat, made so by using all resonant systems, in each frequency band-and sounding terrible! Why? Recall the tweeter illustration we gave above; the reason it disturbs experienced auditioners-and helps sell speakers to the innocent-is that the peaked resonance creates a time distortion, which draws immediate attention to itself-either a painful reaction from the experienced listener, or a delight at the "highs" from the innocent.

So, at any frequency, a resonant, ringing system draws attention to itself, even though the frequency response is "flat". A. R. Bailey, in his famous article on transmission lines, back in 1965, tells how a ringing, sharp cut off, as in classic bass reflex, destroys musical values-but is preferred as having a "heavy" bass characteristic by many. He shows by actual photographs the "spark" response of his system, as compared to the carry on response of the bass reflex cabinet-and then comments, "a heavy bass effect that some people prefer; it is, however, not natural...the author has noted, however, that music listeners were very impressed with the result."

Right in the middle of our discussions about the proper frequency response, we run into "phase". We have seen an overwhelming, somewhat naive preoccupation with "flat" response at any price (even with claims amounting to fraud, at law); when the truth is that "flat" response is desirable, if obtained by non-resonant means. Otherwise, various roll offs at the frequency extremes can be justified, re speaker accuracy, in preference to resonant systems that produce "flat" response. Indeed, much serious work has revealed that a smooth, controlled low frequency roll off is much preferable, in a given size limitation, because the human hearing mechanism can reconstitute the missing or diminished fundamental, provided the harmonics are present in the proper amplitude and time. Again, time enters; and research on how we hear- and, again, the author wishes that more designers and commentators would study their sources, or, as we have said before, do their homework.

We have now restored to their proper places and proportions the subjects of harmonic distortion and frequency response. We have talked of "non-linearity" as being more important than harmonic distortion. As is our want, we now ask, "What is linearity?" It is a very simple relationship, between volts into a loudspeaker and decibels of output. A "linear" loudspeaker, measured statically (we will cover the much more important transient measurements later) is that one whose curve runs at a forty five degree angle, with the input volts to the speaker plotted as the horizontal acoustic ordinate, the output plotted as the exact output. Any deviation from this is a lack of linearity. Unfortunately, very few loudspeakers are linear, on this static measurement. Remember Ben Bauer with his "oil can" effect, and the BBC with their "non linear" characterization of sealed enclosures? On a linearity curve of such systems, the line will go horizontally until a threshold is reached, when the line will start up at a 45 degree angle, later of course to flatten out at the top (this is true of all loudspeakers, the exact output depending on their linearity, this top flattening), when the system "saturates". Thus, we can look at the curve and determine the dynamic capability of a system, and its relative "linearity" on sine waves.

The author knows the above concept is a bit difficult to comprehend, and perhaps totally new to most readers; however, it is a vital part of a loudspeaker, its linearity, probably more important than its frequency curve, and has lead to all sorts of misconceptions about hearing acuity, speaker placement, choice of speaker type, etc. At many speaker seminars, conducted by the author all over the world, the author typically finds attendees who are unhappy because the loudspeakers they own have "no bass" at low room levels; but seem "fine" when played "loud". He explains the above as the primary reason why they are unhappy-which may make them happy, or at least give them a direction for future improvement in their lot!

Perhaps in the future, manufacturers will furnish linearity curves on their speakers, at all relevant frequencies; in place of the amounts of data they do furnish which have little to do with accurate listening. Before we leave this subject, other types of loudspeakers should be mentioned, which have linearity problems, with comments on the categories: all "low efficiency" speakers (we shall show later a correlation between higher efficiency and "dynamic transient linearity"), all known electrostatics, particularly those which are "single ended"; all known deposited ribbon loudspeakers; all dynamic speakers using sharp cut filter networks; all speakers with inbuilt "limiters", to protect low power handling drivers (such as six watt tweeters). We could go on and on, but you must have the idea by now. Of all current U.S. reviewers, only Richard Heyser, of "AUDIO", even mentions this subject; amusingly, most erstwhile speaker designers tell the author that they "can't understand Heyser, nor does the public". So be it.

Other tests which do relate to "linearity", we mention in passing; inter-modulation, Doppler distortion (also known as "FM" distortion), cross modulation distortion, CCIF (first order difference tone-a very critical test, this), Heyser's tests. Later on, we will return to time related testing procedures, which are really similar measurements under the conditions of music. However, before we pass to the all-important time test procedures, we must comment that: In the past, the high efficiency proponents weren't all wrong - their speakers were more linear. You must all remember the famous advertisement of a “horn” manufacturer, which contained one "horn" matched against a multitude of sealed boxes and proclaimed the superiority of the horn. That was true – it was much more linear!

Thus far we have covered various static measurements of "distortion" and commented on their significance and relative weightings re our to be developed or purchased accurate loudspeakers. In the midst of it all, the "time", or "phase" aspect reared up. We have already seen that earlier writers (the author and co-workers) postulated that the difference between their wide band loudspeaker and others must have been time, or phase – both of which, depending on your inclination, are descriptions of the dynamic performance of a system, as it attempts to transduce musical wave forms, constantly changing in every conceivable proportion, but in both amplitude and time. We are now seeing, today, research that indicates most loudspeakers even change their frequency response on music! (We will return to this theme).

So, we want to investigate and reveal data that will help us obtain a loudspeaker for accurate reproduction.

A few years back, the phrase, "linear phase", suddenly appeared in loudspeaker advertising, along with a plethora of loudspeakers with woofers jutting forward (one of them was laughingly termed, by our English friends, the "pregnant kangaroo" speaker), steps on the front baffle plate, tweeters hanging in mid air, etc. Just as suddenly, most of them disappeared, when they "didn't sell".

Most of them were actually horrible to hear, for various reasons; so that the author began to hear from enthusiasts that "linear phase speakers don't sound right". Most of them soon disappeared, to be replaced with another market ploy. Which left the field open to the serious researchers and their companies, for an investigation, with tools becoming available, of the need for "linear phase", "time aligned", and "coherent phase" techniques in loudspeakers.

We were really entering a whole new investigation; we were seriously inquiring into the nature of a musical tone, and how loudspeaker design might be improved to improve the reproduction of music through them. So, we might as well describe the nature of music, with emphasis on how the human hearing mechanism perceives it and evaluates it.

A great deal of independent research has established that the timbre of a musical tone is determined entirely at the onset of the tone, by the particular combination of fundamental and harmonics of the original; so that, if the original combinations are recombined by a loudspeaker in the proper amplitude and time, the timbre of that particular instrument, be it piano, voice, oboe, percussion, violin, or what have you is revealed. Thus, it is that an accurate loudspeaker can even help us determine what kind of piano, what kind of violin, the method of playing (be it American, English, or French)-and even who the artist may be! With that fact now established and agreed to by all the researchers in psychoacoustics, and even by some of the speaker designers, we can proceed. As a note: the author for years has conducted seminars on "the sound of the strings", detailing and demonstrating the varying string techniques of various of the world's great orchestras-this to groups which frankly expected dissertations of tweeters, enclosures, etc.

Now that we know how timbre is detected and evaluated by the listener, we must make it clear that the nature of the propagating system, whether it be electrostatic, dynamic, ribbon, ionized air, or what have you, is unknown to the input signal. The only important consideration is that the transducer be able to reproduce the fundamentals and harmonics in their proper relationship, even if all the data is time delayed evenly. If you believe that electrostatics are always superior, because they are "fast", you need read no further, because you will be disabused. For, it doesn’t matter, within some reasonable bounds, how long the data is delayed, just so long as it is evenly delayed ("group delay") -there is no way for the human to say that the delay changes the timbre, or the artist! Indeed, the phonograph record is the classic illustration, of a group delay that may take weeks, even years.

So, let us proceed to measurements that have something to do with time, for that is what we are now considering. All of you have heard of and seen tone bursts, one of the early and relatively unsophisticated tests of time response. Unfortunately, in any loudspeaker, you can find some region or other which has good tone bursts (the author knows, because such were found in some of his earlier designs, to good marketing avail). But, to get a total picture of a loudspeaker with tone bursts, you must do every frequency, in particular the "trouble" areas, such as bass resonance, crossover points, midrange; and you must do every frequency at low signal, medium signal and high signal inputs. Since that series of tests on one loudspeaker would take a lot of time, years, possibly; the tone burst test is really not our cup of tea-particularly when we also have to compare the time durations of every one of the plots to every other plot, etc.

Back in the early 1970's, the "impulse test" (remember Bailey's spark impulse test, of 1965?) became popular, with Heyser, Fincham, and Leedham (ofKEF) as its principal proponents. But well before that, as we have seen, the principles were well established. Indeed, the author well recalls his primitive impulse test (you flick the stylus and listen) and the effect it had on a well known reviewer; who, when shown the test,

with the statement, "Now, you can in a few seconds know virtually all you have to know about a particular design", went into virtual shock, mumbling that it was just too much, and it took out too much of the mystery.

Be that as it may, you may wish to shock excite your set of loudspeakers, being sure
to keep the level down (too much can blow up amplifiers and drivers). After a bit of experience, you, too, can rightly be called an "expert"!

Now, the laboratory equivalent is the impulse test. In the KEF format, a 20-volt signal is applied to a loudspeaker for 10 microseconds, and the speaker output displayed visually, on an oscilloscope. The exact voltage and time span determines the amplitude and pass band characteristics (by changing one or the other, certain manufacturers have shown in their advertising "impulse" tests that completely ignored the response below 500HZ where most musical energy is). By examining the reproduced shape and duration of the reproduced impulse, particularly the “tail” off, much can be learned.

Proponents of the impulse assert its importance in development, in ascertaining the "attack" and the relative presence of "delayed response", the carry over, which is the principal cause of loudspeaker "coloration” – i.e. "warmth", "coolness", the unique sound of that speaker-which is the amount of response, and the kind of response, it has, when it shouldn’t.

Another test in the time domain is the "step function" test, particularly espoused by the Danish researchers we have mentioned. The "step function" is half a square wave. It is injected into a loudspeaker, and the output examined for congruence to the input. A good loudspeaker will reproduce a reasonable approximation of the input; most loudspeakers do not.

These tests are primarily for the development engineer; they certainly do help him to find out what is happening; and, if conditions are optimal (the market, the "bottom line" accountant, the advertising copywriters), they do lead to improvements. The impulse proponents like their test because it can show the entire response of the system in amplitude and time-indeed, the frequency response of the system can be drawn from it (this is called "fast Fourier transform").

Naturally, they state that the step function test does not necessarily reveal the suitability of a speaker for use in reproducing music. The step function proponents insist that their technique does show the suitability. Interestingly, the impulse proponents concentrate on the "tail off” in time; the step function proponents emphasis the "attack" of the system.

For our purposes, we like both, but we do fasten on the step function, just because it starkly reveals the part of music on which the human hearing mechanism fasten, the leading edge.

We now leave the tests, which do reveal dynamic distortions, and proceed into the mysterious world of "phase", which we have been touching from time to time, in unexpected ways. Remember the wide band loudspeaker of some years back, when we more or less guessed that it must be more phase proper than other loudspeakers?

Phase is merely another way of expressing time relationships; in our subject, loudspeakers, it can be a graphic illustration of the differences in time of transit, from the input voltage to the output signal, charted as a function of frequency. Enough of that – is phase important?

The BBC seems to contradict themselves; some early work by the now legendary Shorter was in the time domain; later statements in their work indicate its relative non-importance-unless you listen in stereo.

Some well-publicized listening tests <u>seem</u> to show a preference for random phase loudspeakers, or no preference when listeners are offered a choice between random phase and more linear phase loudspeakers. Many loudspeakers currently being offered for sale are praised to the skies by some reviewers, which loudspeakers are by design incapable of passing a recognizable step function.

Well, is phase important? After collating all sources, we emerge with the following rather self-evident statements:

1) It really is not important in most loudspeakers, which have serious problems that overwhelm considerations of phase.

2) It is really not important to the vast majority of listeners, who are casual re: the relationship of reproduction to the "real thing", live music.

3) It is the distinguishing mark of a really accurate loudspeaker, and its results in sound reproduction are of extreme importance to those who by conditioning, training, and preference are sensitive to musical timbres. Again, we ought to make these divisions meaningful, by quoting "true incidents from real life", as the pulp magazines once said.

Several years ago, the author entertained two young visitors, one a highly trained professional violinist, the other an equally trained baritone (the violinist an American, the baritone a Frenchman). They related how, together, they had conducted an extensive search, for some seven months, in and around New York City, for an "ideal" loudspeaker; and that they had settled on a large monitoring loudspeaker of the author's design. The reason why they had come, they said, was that they wanted to tell the author why, that only his loudspeaker, of all they had auditioned (all the famous brands), reproduced the actual timbres of their instruments, and, of course, the singing voice. The author, being a bit of a skeptic himself, questioned them closely, and played for them less complex designs of his. They were very consistent--only this large, complex speaker pleased them, and they were rather scathing of the author's lesser designs, perhaps a bit less so than some other loudspeakers then available.

Some two years later, the young violinist returned from his concertizing around the world, to hear the updated version of his loudspeaker. The author played for him an alternate design, a very expensive monitoring loudspeaker, worked out with the cooperation of the Danish researchers mentioned above, with highly advanced drivers and series network crossovers of first order. The young violinist listened about twenty seconds, then said: "Is there a crossover in this speaker around 300HZ?" Asked why he thought so, he asked the author whether he didn't hear how the string choirs were being segmented! He then voiced a strong preference for his original choice, now updated, because it was, he said, the only loudspeaker that he had ever heard that didn't mess up the string choirs!

These young men, intelligent, educated well (Juillard, Curtis, studies with Sherrill Milnes, principal baritone of the Met) were very phase sensitive! Completely absorbed in making music, they rejected all other loudspeakers with crossovers of any kind in or near the basic fundamental energy of music. Such listeners instinctively prefer more phase worthy loudspeakers, because they are very involved in the timbres of instruments (during the original meetings, they had pointed out to the author how an English horn passage was properly reproduced only by the large monitoring loudspeaker; indeed, they chided the author for designing the smaller systems that couldn't, in their evaluations, reproduce the original timbres.)

Very few listeners are so conditioned. The second loudspeaker played at the second session for the young violinist is a phase linear loudspeaker, highly regarded in its own country; and the design engineers who worked with the author are still shocked that anyone could hear that 300HZ crossover on which they had worked so long! The young violinist had rejected a very slight alteration in phase, which, he proclaimed, "fragmented" the string choirs.

At a comparative speaker demonstration in a dealer showroom, the dealer asked the author to listen to his "competition", a moderately priced dipole, which had received rave reviews in several of the "underground" journals. The dealer played a selection from Tschaikowsky, and asked the author whether he didn't think the sound was "musical". The author refused to answer that query, as he commented, because he didn't know what "musical" meant (see above). On further questioning, "Well, what do you think about it?", the author replied immediately that it obviously had no dynamic range-which mystified the dealer. The author suggested that the dealer find a recording of a piano, and play it first on the dipole, and then the author's speaker. The dealer found a piano disc at the bottom of a stack of synthetic rock discs; put it on, first on the dipole; then on the author’s speaker.

The dealer was shocked, because on the switchover to the second speakers, there immediately could be heard the hard "attack" sound of a piano, the "hammer sound", which was entirely lacking on the dipole reproduction. The author commented that was the "dynamic range" difference; and further suggested that his speakers had not changed the disc, merely reproduced it; that the "musical" (how the author hates that phrase) speaker lacked rise time, resolution, dynamic range, and whatever is involved in the accurate reproduction of musical timbres. The dealer salesman later switched to a very famous "box" with phased array dipole drivers, speaker, saying that this was the ultimate. The author commented that the hammer tone was lacking there, too; whereupon the salesman said, defensively, that "many people like it that way", without all that ugly hammer tone. The author smiled, commented that the precise same incident had occurred six years before, six hundred miles away, and with the same conclusion-some people don't want to hear the music (in this earlier case, the recording engineer who had made the master tape of the piano entered the room during the demonstration and with great glee stated that he had made the recording with the microphone right at the sounding board where there was plenty of hammer tone).

A year or so ago, at another dealer's shop, a fervent summer part timer, who has assured the author that he understood all the advanced stuff because he was in his second year of engineering college, was shocked to discover that the "strange" sound he heard on a large monitoring speaker of the author's design was actually the "skin effect", the initial attack of a jazz drum, when struck with a mallet (he had never heard the drum live, and had to be assured that was correct, by other salesmen present, who played in jazz bands). He was a few minutes later nonplussed, because on a record he had carefully picked for "evaluation" of another, smaller loudspeaker of the author's, against an English "monitor" costing three times as much, two people present pointed out to him that the less expensive loudspeaker reproduced two instruments, an electric bass guitar and a kick bass drum, hitting at the same time; whereas on the expensive English loudspeaker, it was impossible to distinguish the drum as such – there was a blurring!

The young man, when explanations of "attack", "phase", step functions, etc. were given, went into depression, saying that he had been betrayed; and he questioned the "honesty" of the manufacturer of the English loudspeaker.

The author received a letter, detailing the following: One owner of a much earlier design of his went speaker shopping with a friend. The friend was impressed by a slightly modified version of the earlier design, which was "on sale" at a shop in New York City, and left a deposit to hold the speakers. The letter writer then suggested that the potential customer hear a later design of the author's, just to be sure-and even though it wasn't "on sale", so to speak.

They arrived at a dealer featuring the later designs, carrying with them a disc made from a digitally encoded tape, with brass chorales. The dealer played the disc, and they were both very pleased, much more so than from the earlier "on sale" design; but, the customer insisted that, just to be sure, they then proceed to hear a large electrostatic dipole which had been highly praised by most of the underground reviewers, and by esoteric dealers. The writer described what happened: The dealer placed the record on the turntable, turned it on, they adjusting the level to their preference. Immediately, the writer pointed out, the sound just "shriveled up". They spent no more than two minutes, took off the disc, thanked the dealer, returned to the second dealer and ordered the speaker. The writer thought the author should know all this.

A day or two later, knowing the third dealer, the author called the dealer on the phone, and with a spirit of good humor, read the letter to the dealer. The dealer immediately remembered the two and said, that they were trying to play the brass "at 105db." The author commented that brass is much louder than that, when several soloists play together. Whereupon the dealer said that, in the right room, in the right position, and with the right record, the dipole really wasn't bad-and that, besides, he was able to sell a pair a week: Why not?

What lessons might we draw? Proper attack time is not relevant to many dealers, and they insist, to their customers; proper dynamic range, which we will see is consonant with proper attack, (when the fundamentals and harmonics occur in the right time sequence, with the proper amplitudes, the timbre and attack of the instrument are clearly defined) is sought by many. Proper phase and its ancillary qualities is exceedingly important to highly trained and motivated listeners, possibly to those who are acquainted with and relish the qualities of live music.

Is there a phase perfect loudspeaker? No, but some are better than others. We know that phase changes (see the illustration re the crossover, above) are to be deplored. We know that smooth phase changes are better received by the hearing mechanism. We even know that absolute phase is audible, under some conditions, though rarely achieved in our current reproducing chains.

We have known for a long time that we are most sensitive to phase changes in the midband, so that any crossovers with rapid changes should be avoided (but see above, the violinist heard a mild phase change at 300HZ). We know that phase changes are audible, though less critically so, at the frequency extremes; as compared to a loudspeaker that has less phase change (remember our ideal loudspeaker of some years ago?). We know all this, and our illustrations confirm that some people prefer to hear a less altered musical program; whereas, and this we feel is bound to change as the consumers become more informed and critical, it is now easier to "sell" an altered program speaker to most people than to struggle through the painful educational process that is involved in helping people to understand that pianos have hammers, that brass does play loud in chorales, that drums and electric bass guitars can be separated in real life-and should be by loudspeakers.

Well, we have covered our introduction, and the test procedures that help us distinguish accurate loudspeakers from entertainment speakers and altered acoustic analog speakers. Can we now construct a loudspeaker that is ideal, for this stage of art; i.e., more accurate than that early one we have mentioned; or, at least, a near ideal loudspeaker, by definition one that has gradual phase changes only?

Yes, we can; for the tools are now available to construct a basically aperiodic loudspeaker, a speaker without inbuilt periodicity (i.e. resonance), not necessarily a perfect example of aperiodicity (perfection is always in the future), but much closer than the hi-fi market generally offers. We have the test procedures; the requirements have been set down in some length, if not detail, above. Our task now is to get the parts working properly. But first a story, which illustrates the situation: Some years back, "computer designed" cartridges were introduced to the marketplace. The author was at that time working in England with associates on cartridges, and, sadly, had no computer at hand. One of his associates wryly commented that, after the "computer design", you still had to find the piece of rubber hinge that did what the computer "called for" - so let's get on with finding the right piece of rubber.

The same is true in loudspeakers. So, let's get on with finding and assembling the right parts. What we have to find, first, are "minimum phase: drivers (there's that pesky "phase" word again-the bane of many designers), which simply means that the frequency response and the phase follow each other, so that if you vary one, you find that the other follows closely. Our loudspeaker perforce will contain only "minimum phase" drivers.

Just to illustrate, let's give examples of non-minimum phase drivers; which is simple, because any resonance (remember the tweeter illustration) leads to non-minimum phase. So, let's go with obvious examples: a) Electrically-resonated systems, such as electrostatic bass panels known to the author. Peter Walker, in his original articles on his electrostatic, gave a "Q" figure of three for the resonant uplift in his speaker, stating that wasn't as bad sounding as it looked, because of "room involvement", and the ear's comparative non sensitivity to resonance below 50HZ or so (Raymond Cooke, of KEF, repeats the magic figure of 50HZ in various of his writings, that you don't have to worry too much below that). b) All dynamic drivers with low magnetic control, or "force factor", as designed for use in "commercial" loudspeakers (Lian's phrase). It turns out that large magnetic control systems are less "resonant"-magnets are good things, though expensive. c) All drivers which use resonance, "decoupling", or "breakup" (you use the word, decoupling, if you want to impress the customer with your engineering skill, the word "breakup" if you want to affirm what is actually happening) to "extend" the response upward.

Just to be more specific: Let us suppose that you own a two way loudspeaker, with an eight inch bass-mid driver, a crossover at 3KHZ, and a treble dome. Chances are you might; there are many such around. Let us ignore possible resonant effects from the crossover (we will cover that later) or from the enclosure design (also later); we are talking only about the drivers. In order to carry an eight inch cone up to 3KHZ, unless it is so designed that it has a very narrow band of output right on axis, you must have resonance, decoupling of the cone, or whatever you wish to call it. Cone breakup is "non piston behavior", and thus is not "minimum phase". So, the first thing we can say is that, even though you have seen a smooth response curve on the speaker in that region, the reproduction in that all important midband is only vaguely similar to the input program, in time. Even though it may sound "smooth", or "musical", or "having great depth" to you, or a reviewer, it just isn't accurate-indeed, we might conclude that its lack of accuracy can be interpreted as the above adjectives (remember our "musical dipole"?).

Your dome tweeter, unless it is one of the very rare, expensive, "aperiodic" units developed in Denmark by the people we have named so many times, undoubtedly extends the treble response by dome resonance effects, more or less damped by the designer. However, this kind of resonance is nowhere near so pernicious as the bass-mid resonance effects, because the ear is less phase sensitive up there, and the audible effects can be minimized with such description as "hard", "sparkling", "true highs", "disembodied top", or "musical" depending, again, on how you respond to this phase distorted region.

The same rules apply to electrostatic, ribbon, deposited conducted drivers, etc.-changing the material may or may not influence the effect, or may, depending on the design skill. But no particular form of loudspeaker drive is inherently less resonance prone than another-it is all the way the drive is designed and used, as we will illustrate later on.

So, we now want to throw away all those resonant drivers and build our loudspeaker from "aperiodic" drivers. We take out the bass-mid unit and we replace it with an expensive driver made in Norway, which has a patented "dynamic damping" magnetic system (remember how important magnets are?), that prevents the driver from running away with itself, at high signal levels, in bass reflex cabinets (many of these are around, particularly in BBC influenced designs-see above). Will that help? Well, yes, it makes the bass much crisper, but doesn't affect the midband.

So, now we go back to the manufacturer, and tell him we want this expensive driver, but with a "better" cone material, i.e., one that has less breakup effects in its piston area. He supplies the driver with the new "miracle" material, polypropylene, and we find immediately that the piston range is vastly increased, and we are happy, until we find that the cone dives sharply well before 3KHZ, leaving an enormous "energy hole" in the speaker. So, we fiddle with the crossover, etc., trying to make this better cone go up to 3KHz-which it never does.

Similarly, we replace our resonant tweeter with an aperiodic dome from Denmark. We put them in place of the original drivers and feel content-now we are non-resonant-but we are also non continuous-the speaker has violent changes in energy, directivity, even phase. So, we are stumped, and give up. Or: We decide to develop specific drivers for minimizing the problems in our loudspeaker; we have illustrated the two-way problems, which have been minimized in practice by a few manufacturers working with new technology. The three way problems are just as bad, probably worse, as we shall see when we get into crossover design, later.

The drivers we develop do have new cone materials, such as composites, certainly not the polypropylene, paper, coated paper, and bextrenes of the past. They are also quite sensitive, or efficient, whichever word you prefer; because they have a high ratio of magnetic strength to cone mass (BL factor, force factor). So that the speaker we will develop finally will be much more sensitive than the high quality designs of the past; this being all to the good, because we want this new speaker to be very linear, to have great dynamic range and high instantaneous peak power output-for reproducing the coming digital programs.

These drivers also have an interesting characteristic; each has a relatively uniform impedance characteristic; because high magnetic strengths tend to restrain the traditional rise in inductive impedance (no point in explaining that now, only to say that it exists); and thus to produce a smooth frequency characteristic, which slopes off naturally beyond the region where we plan to use it. We are done with "flat", resonant drivers; we are into the era of "aperiodic" drivers.

So we can now join our drivers with a crossover. We want to use a crossover that will permit the system to have a sharp attack; we want the woofer, the midrange, and the tweeter to be joined in such fashion that they can, together, reform a step function. We also want to present a primarily "resistive" load to the amplifier, because "reactive" loads tend to store and discharge energy; and that is not what we want; we want the electrical energy to get through to the drivers when it should, not later.

Interestingly, the text writers uniformly agree that, with "good" drivers, the "first order", six db/octave networks are the only way to go. New research, in England primarily, indicates that "compensated 12 db/octave networks" (with slope rates of less than 12db/octave) are a satisfactory second choice. So, if need be, we will use only these as a second possibility; and if we must use a sharper slope (12db/octave or more), to protect a driver, we will do it only on the "high pass" (the upper frequency driver of a crossed over pair) side, because, again, the degradation of transient performance is less when a driver is sharply cut off electrically at its bottom limit, than if it is sharply cut off electrically at the top of its limit.

Both "first order" and "compensated 12db/octave" networks are useable. Why only these two, we knowing that many, many loudspeakers incorporate sharper networks? The answer is our fundamental one; we wish to produce an aperiodic loudspeaker that can reproduce the onset, or leading edge, of instrumental data; so that we can hear the actual timbres of musical instruments; and this requires a crossover that will pass the amplitude and time domains properly-which our two types will, and which no other type will!

Most of you will find this hard to believe, because you have been reading long advertising claims about "phase compensated", "third order damped" crossovers, etc. So, let's go back to basics.

Many loudspeakers use 12db/octave networks. To do this, and to maintain flat response at the crossover frequency, the two drivers must be reversed in polarity, so that one pushes and the other pulls, on a given signal; in addition to which, the two signals (one "positive", the other "negative") from the drivers are time displaced, i.e., they occur, first one (the upper), then the other (the lower).

Can you hear this? Those who design such speakers must insist that you can't. Others insist that the polarity reversal of the two drivers produces a "hollow" effect. The author insists, after years of comparative listening, that you hear the effect of the one with the proper polarity (two the same) as having greater dynamic range, greater "attack", greater homogeneity, and a sense of listening ease. We might mention a recent development project at JBL, described by Peter Moncrieff in his IAR journal, to wit: He heard a developmental prototype of their Model L250 speaker at the factory, and told the "assembled JBL design team" that the music lacked convincing "coherence", all the frequencies being there but not "jelling" into real sounding instruments. The prototype had second order (12db/octave) slope crossovers, for driver protection, he was told. Later, the design team, including John Eargle (a famous name in professional audio), tried first order slopes and reported that the sound of the L250 got dramatically better, music sounding more real to them. Even record ticks sounded very different (we can predict that from our discussion above of how higher slopes displace the energy in time), more coherent and neutral!

A few current "monitor" speakers employ even higher slope rates, which do permit proper polarity in the drivers, but which introduce even more problems of transient instability, particularly with extremes of input. What the JBL design team noticed, and what Mr. Moncrieff reported on, must by the very nature of the crossover be made worse with steeper slope rates-all the text writers mention the danger in them.

Crossover design is not a magical, mysterious science. The rules have been laid out, and we will adhere to them in our near ideal loudspeaker.

We must mention "active" crossovers (ahead of the amplifiers), bi amplification, and such; beloved of the "esoteric", "freak" fringe of the industry. Active crossovers can be useful in joining together loudspeaker elements that are otherwise poorly matched, vis a vis impedance, sensitivity, and resonant systems. Their use, however, has no effect whatsoever on phase, our primary concern. While the author has heard several laboratory engineered "active" systems, which are quite good, having taken into account the time function among the drivers, he has also heard colossal abortions, with woofers separated great distances from mids and tweeters, the all brought together and made to work by "hi amplification", or so the proponents assert.

Otherwise, all active loudspeakers suffer from the very great expense for results achieved, as compared to loudspeakers engineered with modem technology and passive crossovers. They are obviously tricky, subject to the geometric increase in breakdown possibilities (if you add one more electronic gadget, the probability of problems goes up geometrically, not by two, but by four); and very hard to service. Therefore, we dismiss their use in our ideal or near ideal loudspeaker.

So, back to 6db/octave, if at all possible, and compensated 12db, as an alternative where our drivers just aren't good enough for 6db./octave. Does this conclude our discussion of crossovers? Not at all; we are just getting into the meat of things, as faced by the design team we are becoming, to develop our loudspeaker.

Many of you, possibly all of you, have seen the impedance curves of loudspeakers, which typically resemble mountain ranges. For instance the famous LS3/5A BBC mini monitor has an impedance curve that ranges between eight ohms and sixty ohms. Its impedance curve reflects a very complex series of circuits designed to equalize response (including the "BBC dip") as well as to join the drivers. The author and his co-workers in England, a long time ago, concluded that "flat" impedance curves were much better for the sake of power transfer, for "linearity", and for dynamic range. So, the BBC theories went right up against the theories of the author!

You recall that the BBC more or less invented the equalized crossover concept, that, not believing in phase in those days (indeed, not knowing any more about it than anyone else), they employed the crossover network to flatten the response on axis, i.e., to flatten it into the BBC desired response curve; which was done by a complex grouping of chokes, capacitors, resistors, and auto formers (mating transformers).

All British speaker design was influenced by the BBC, so that a whole generation of English speakers contained crossover networks equalizing the response curves. The English testers and reviewers, in their graphs of frequency and impedance, always attached to the latter, "This is not the frequency response". The text would always comment that, since the impedance did not fall below a given figure, the impedance was correct.

Today, the current thinking is that the impedance curve influences the "dynamic" frequency response, meaning that, under the violently fluctuating conditions of music reproduction, the frequency response of the speaker may be constantly changing! In addition, many authorities say that the impedance curve variations, if any, must be examined for indications of non-linear or phase distorting conditions. We will also point out later that a sharp rise in the bass impedance can be correlated with extreme room interactions, when we discuss enclosures.

The author has long believed that, the flatter the impedance curve, the better-this even before the author and his coworkers knew the reasons (see the ideal speaker, above). The Danish researchers are in full accord, they having examined the step function changes as one permits the speaker to have an erratic impedance curve.

Today, the BBC seems to have abandoned its ragged impedance approach, since the current BBC Monitor is actually a bi amplified two way system-with no passive crossover, all equalization being done ahead of the amplifiers.

For the skeptical, we should explain why flat impedance curves are desirable in our ideal or near ideal loudspeaker. When an amplifier "looks" into a high impedance load, particularly if that load has high "reactive" elements, it will not deliver its rated power output, it may go into "protect" at output powers which are only a fraction of its rated power; or it may "clip" at a fraction of its power capability (which is always referred to a resistive load). When an amplifier clips at any frequency, it clips at all frequencies; indeed, this characteristic is an argument recently advanced by a bi amplifier addict when the bass amplifier clips on a high impedance bass peak, it won’t blow the tweeters!

The original BBC thinking was restricted to "small signal" conditions. Modem thinking on crossovers encompasses "large signal conditions". Our thinking, long ago, on our then ideal loudspeaker was to maintain flat impedance by doing the driver equalization and matching by inserting resistor-capacitor networks, rather than by using larger chokes to equalize (while producing rapid impedance rises to peaks).

With the better drivers that are available or can be developed, today, equalization networks may be used less, and the load made even more predictable; so that the amplifier, the phase, and the performance are more predictable.

Again, just to show how this works: Some years ago, at a dealer seminar, the author was introducing a new, large loudspeaker with relatively flat impedance. The dealer had supplied a Sheffield direct cut disc, which was merrily playing at a very high level on this new loudspeaker. Someone in the audience asked to hear an earlier design, a smaller loudspeaker; one that had actually been "computer designed", with a very elaborate equalizing network, 18db./octave filters, and a ragged impedance curve. The author pushed the button for that speaker-and all the fuses blew! The dealer asked why (it had happened before). Now you know why; the complex crossover in the earlier loudspeaker produced an impedance curve that the amplifier (a very high powered Japanese unit) did not like; the amplifier "clipped" and the fuses were blown.

Therefore, our near ideal loudspeaker will have a very flat impedance curve. We will relate the curve to the enclosure later on; for now, the crossover will produce a very flat, resistive match to the amplifier, so that we can hear the dynamic range of the best program-no blown fuses.

We still aren't done with crossover design. Recall the young classical violinist illustration, re "phase" audibility; the young man who immediately heard a series connected, first order network at 300HZ in the new design; and thus preferred the updated older design with the 100HZ first order crossover. The BBC, which was rather inconsistent in their views on "phase" in the past, did recommend that all crossovers be kept below 250HZ and above 2KHZ, to avoid the phase critical midband regions, the region of maximum hearing sensitivity. The author now believes that the lower crossover in a three-way loudspeaker must be kept at 100HZ-and in the series connection.

At a recent industry conference, the author presented an updated version of one of his monitoring loudspeakers, using new drivers and a 1OOHZ crossover (slightly lowered from earlier versions of the same), first order. Next to it he presented a new loudspeaker, using virtually the same parts, with a slightly simpler bass enclosure; the only real operating difference between them being that this new loudspeaker, being built all in one, incorporated a series connection between the woofer and mid drivers, first order, at the same 1OOHZ crossover point. Virtually every critic who heard the two, side by side, preferred the new design, on singing voice and piano, which we already know are the two most difficult signals to reproduce. When the author commented that they were hearing the difference between a parallel connection, first order network at 1OOHZ, and the series connection at 1OOHZ, they gulped-because the implication is there, that every other point and rate and hookup would be much worse in distorting what they heard. What they liked in the new design was the absence of a "ghost" quality (we can predict that, from a slight difference in arrival time), a greater homogeneity-and more dynamic inflection. One critic every said the newer design showed less "record breakup"-as if we had changed cartridges, he thought!

So, our near ideal three-way loudspeaker will contain a first order crossover, at 100HZ, and with the series connection-because every other slope rate, frequency, or connection will materially degrade the performance.

We are finally finished with crossovers, impedance and the influences on what we hear. Naturally, if we don’t want to involve ourselves in the tremendous cost and size of the parts required for a very phase accurate crossover, in our three-way system; we can just design a two-way system. And for this reason alone, most two-way systems are superior, on a listening basis to virtually all three-way systems!

Finally we proceed to the enclosures for our ideal, or near ideal loudspeaker. We must consider enclosures for the bass drivers, the mid drivers, and the treble drivers-believe it or not, this last is important, too.

Some time ago, the author delivered a series of lectures to various professional gatherings, concerning the bass of music, its natural reproduction, and the theories surrounding. We have already mentioned the lecture to the BAS. The one we will now discuss, the most interesting of the lot, was delivered to a professional and amateur gathering in Paris, in French, which is a difficult language for technical discussion, "De la basse de l'Orchestre" (concerning the bass of the orchestra). Present were authorities from England, Scotland, the U.S., Germany, France, Spain, indeed a cross sampling of many approaches to the art of the loudspeaker. We must understand the nature and background of this group--because the author's observations were not challenged, except as to particulars (the inventor of the Linn-Sondek turntable meekly requested a clarification, in English, on one of the phase diagrams offered on the blackboard).

The author listed bass enclosures in order of descending quality, as follows: 1) Seventeen foot concrete horns; 2) Transmission lines; 3) Acoustically damped duct systems, which were termed "line tunnel" enclosures; 4) Bass reflex; 5) Air suspension. After a short lecture, the ensuing discussion period was fascinating, because there was general agreement-the author's rankings were accepted by everyone. The author commenced with a discussion of the nature of music, which, since the "Thorough bass period", of the Seventeenth Century, has as its very foundation the bass; then wryly suggested that most speaker designers seemed intent on changing history-and pretending that the proper reproduction of the bass was not so important, or as sales worthy as a new kind of tweeter, or some such. The audience did laugh at this, a bit nervously, to be sure; because they were jointly indicting their own loudspeakers as not being designed for the reproduction of the nature of music. We want to reproduce music for our ideal or near ideal loudspeaker; so let's summarize that and similar lectures:

1) Seventeen foot concrete horns, with 80HZ crossovers, once heard, will never be forgotten. We mention these lovely devices, found in England, because the reason they are so memorable is that they move air in a linear fashion, without resonating; are thus incredibly accurate in both amplitude and time-and sound it. They are huge, rather expensive, not portable, and will not be sold in great quantities; but they do move air properly, without resonating, rather, by having a proper "acoustic radiation resistance". Please remember this phrase, because it explains all that should be known about bass propagation, as we shall see below.

2) Transmission lines. Much discussed, hardly understood, severely criticized by some "authorities" who haven't done their homework, and who insist they are really "reflex", or "non-optimal", or "high distortion"; these devices are closest to the great horns in terms of their "acoustic radiation resistance". Now we must explain just a bit about acoustic radiation resistance. It is shown in all the texts as a curve of efficiency vis a vis frequency, of a given sized diaphragm; so that the response will drop off below the given frequency, unless cone velocity is increased, i.e., <u>unless the system is resonated.</u>

It is the great advantage of the transmission line, this increase in acoustic radiation resistance. In the horns, a taper and a large mouth opening govern; in the lines, the mass of the diaphragm is placed in parallel with the mass of air in the line, forming an "equivalent acoustic air mass" such as would be moved by a multitude of perfect drivers in a wall--or by a large horn. The resonances in a line, properly engineered, are all just below the program range so that they delay the impact signal less than if they were smack in the middle of the program. A proper line thus has what is called "zero ambient pressure: no pressure build up to hamper the "attack" of the driver. Lian, of Denmark, comments that this is the reason why musically trained ears always admire the quality from lines-the quickness of response. As we now know, that determines timbres.

3) Acoustically damped duct systems. These lack the acoustic multiplication of horns and lines. However, they are otherwise similar to lines. They do double the source area of the extreme lows, and can be adjusted to give near aperiodic conditions in the audible band. Adjustment can be done on pulse, step function, or impedance tests; the object being to get the best possible balance between frequency response, harmonic distortion, and "attack-decay" characteristics. Interestingly, the best balances between the above factors occur when the impedance rise is limited to 3db! There are other variations on this basic idea, of applying a heavy damping to an opening in the enclosure. The Danish researchers call the opening's effect a "flow-resistance, damping the resonance like a DC-resistance in the oscillating circuit...gives more a precise bass response and better woofer quality...the impedance maximum is reduced at least 50%...the amplifier is able to give more power in the lower range." Another variation is the author's "pressure release". All are for the same purpose; by controlling the acoustic environment of the driver, it is made to be more "aperiodic".

4) Bass reflex systems. There has been a movement back to them, occasioned by the need for greater sensitivity, and by the popularization of the designs parameters of Thiele and Small. We have already mentioned one of their gross problems-at high signal levels, as the driver leaves the center part of its gap, it becomes un-damped and just rolls on. Special magnet systems (we have mentioned the patented driver form SEAS) can minimize the effects, but are very costly, and thus hardly ever found in production loudspeakers. Another problem they have is source impedance-all the formulae predicate a zero source impedance from the amplifier, which never exists in real life. As soon as connectors and wire get between the amplifier and the speaker, all the formulae go out the window-and the speakers roll on! The entire cult of special connectors and wire rests upon this dubious premise that a speaker requiring zero source impedance can be made to function in real life.

5) Air suspension. By now you probably know that the author and his associates look upon air suspension with near hatred; because it is the most used and least time accurate, of all enclosing systems. These are high-pressure systems, rejected by the BBC for "non-linearity" at low levels, by the Danish researchers because they produce "non-linear time delay distortion". So, we pass them over, because we want a time accurate loudspeaker.

Thus, our time accurate loudspeaker, our near ideal loudspeaker, will use a transmission line bass, if space and cost permit, and some type of acoustically damped duct system as a second choice. Before we go on, we must consider the reasons again-time accuracy. In the early days, during the development of the ideal loudspeaker in England, the author and coworkers found a fascinating effect- a proper enclosure could be ruined by one misplaced piece of damping material obstructing flow away from the driver. The effect on "noise" was shocking-you could hear the early reflections bouncing back at the cone and dulling the attack. One coworker said the difference was between "crisp and clear", to "strangled and stuffed"! This illustrates well how seemingly minor aspects of speaker design become major, when the object is to create a time proper loudspeaker. It is as if all factors conspire to reduce our efforts back to the level of the time improper stuffed box speaker of our childhood, the jukebox, the conditioner of all too many speaker preferences! It also explains why so many designers ignore the proper time design of enclosures-it does require thought, care, and extra expense.

The midrange has the same problems, only made worse by the fact that we can hear them more readily. The best mid enclosure is a long, damped tube leading away from the driver, with "flow-resistance" at the other end; adjusted for aperiodic damping, just as we did for the bass driver. Next best mid enclosures are irregular spaces with "flow-resistance". Worst of all are the stuffed small chambers behind most mid dome units; indeed, only one manufacturer of domes today makes any attempt to linearize the time response, he installing a double chamber system behind the dome. The very best mid dome the author has ever seen and heard was an experimental unit, made by KEF, back in 1967-which had a long, tapered flexible tube attached to it. It was never manufactured, the reason being "cost". Our query-is cost always to govern our ability to hear time proper musical reproduction?

Similarly with treble units; Arthur Radford, one of the pioneers in lines, long ago postulated that a proper treble unit would have a miniature line attached to it! To date, no one has; though the Danish researchers we have mentioned time and time again do employ double chambers to minimize the phase destroying backpressures.

We have now covered the design criteria for our ideal and near ideal loudspeakers. We have emphasized again and again proper design for the time domain. We now know that a speaker designed with such emphasis will also have a fairly wide, smooth amplitude response; whereas a loudspeaker designed only with regard to "flat frequency response" may have a horrible response in time (refer to our resonant illustration).

So, are we ready to make the ideal or near ideal loudspeaker? Not at all, because we still haven't covered the data that determines whether an auditioner will be able to experience a close simulation of what he hears at a live performance. We still have to consider the interaction of the loudspeaker, the room, and the perceptions of the auditioner. We must determine how this affects the geometry of the loudspeaker; possibly it's kind of drive systems, and its radiating characteristics. At the end of the possibly long trail, we are faced with acoustic necessity, and listener acuity.

First, the shape of the final loudspeaker: Are broad source reflective devices useable, or unitary dipoles that offer a broad source (a description that does not cover the Quad ESL 63, which is a "point source" dipole), or various "omni" configurations?

The best way to determine the shape is to return to basics, to "first principles", if you will: Our goal is to make a loudspeaker that will reproduce the actual timbres of musical instruments, in a room, and to a listener. If you think about that goal, we really don't have much choice at all left to us, as we will see.

We have developed as our first possibility three drive systems, in time proper enclosures, joined by a network that makes them "phase coherent". These we must get to work in a "phase aligned" fashion, i.e., all providing the proper data to the listener at the proper time. The only way known to the author is to place the drivers on a sloping baffle, so that the propagating center of each driver is in vertical alignment with each of the others. If we place them on a flat board, we will be phase coherent but not phase aligned; if we place the bass units out in front, with a step back to the mid, and another step to the tweeter, we will get proper time at just one seat on axis, every other seat in the room being very out of phase; and we will have introduced severe problems in the vertical plane. The best we can do is to slope the baffle, either by designing it in; or by recommending that the speaker be used on a tilt back stand.

But we aren't "home" yet; we still have to do some serious thinking, because we want to preserve our hard fought gains in the drive systems, and to make sure that they aren't wasted in the air, on the way to the listener. So, we have to re-examine how we hear; as well as how sound waves propagate in nature; and we must construct our loudspeaker so that we have the optimum propagation, and the optimum accord with the human hearing mechanism.

We go back into the acoustical texts, and find that: 1) the higher the frequency, the smaller the source should be, for proper dispersion. 2) the higher the frequency, the more it will not pass around sharp comers, but will "re-radiate", i.e., radiate at a time after the original radiation. So, we are faced with the problems of enclosure boundaries, source size, all that.

It is always easy to illustrate situations with gross mistakes in loudspeakers, there being many more such than the laws of chance would permit, in this world. Some time ago, on the West Coast U.S., the author was asked to hear and comment on an audiophile column loudspeaker; which had a broad frontal section, containing drivers of every sort, including a ribbon super-tweeter. Before he heard it, several audiophiles present asserted that the speaker had "no highs"- that drawing the author's attention to the ribbon. At the end of two minutes of listening, the author commented that there was no "impression" of highs, the reason being that the designer hadn't done his homework, placing a source of extreme highs on a very wide baffle; the result being a smearing from re-radiation, of the first signal from the tweeter and succeeding signals from the cabinet boundaries. The author explained that the speaker measurements, on a sine wave basis, might be good- even though the sound was "chaotic".

Why no "perceived" highs? It is necessary again to investigate how the human hearing mechanism perceives. We learn that the human being perceives a single stimulus if following stimuli of the same kind are separated enough in time (that word again), depending on frequency. If the stimuli are too close in time, the human mechanism integrates them and hears a "smear"; if the stimuli are delayed, one to the other, enough, the human hears the first stimulus (signal) and identifies it; the later signals being separated in the hearing process, and identified as something else, such as reverberation in the listening environment. We call this the laws governing "first arrival" and "late arrivals"- and these laws apply to all hearing, all loudspeakers, and all designers.

Now you know the mistakes in the column loudspeaker. You also know why our source of extreme highs must be very small. You now know why unitary dipoles and all other wide source speakers cannot possibly have perceived highs-the ear just can't distinguish them!

As we go down in frequency, we expand our sound source, taking due regard for edge diffraction effects, by rounding the comers; until we reach the extreme lows, which demand a large source, and where edge diffraction effects as such may not be so significant (but are, in another context, see below).

We search for a shape that is in accord with the laws of acoustics and how we hear. We don’t have to search very far; the text writers all describe the ideal speaker shape as an "expanding sound source". Nature’s shape that corresponds is a pyramid with rounded edges. That is our shape for our loudspeaker-so that we can "hear" the musical spectrum.

Please make sure that you have the above facts referring to the way we hear in your firm grasp, because we will later refer to them as the dictators of room positioning for our near ideal loudspeaker.

Well, we now have our "point source" for the highs, and our "expanding sound source" geometry for the complete loudspeaker. We make such a speaker. We want to reproduce "stereo". The next query may seem useless, because everyone knows what stereo is. But do they? Stereo is a well-defined approach to sound reproduction invented by an Englishman, Alan Blumlein, one of the few authentic geniuses ever to work on sound reproduction. He investigated the hearing process, and concluded that the "crossed microphone" technique was the proper pickup, to be reformed in the listening room by two "point source", spaced loudspeakers. He even invented the stereo disc. Now, some fifty years later, the most advanced recording people are returning to his early dictates. But apparently most speaker designers aren't.

Let us ignore the designers who don't study stereo, perception, acoustics, etc. (one authority wryly commented to the author, long ago, that there are very few first rate minds in audio). Let us take our properly designed loudspeakers into a listening room and place them properly.

James Moir, another Englishman who has actually studied our subject, and who has written on FM distortion, phase, and how we hear; recites an example. We take our loudspeakers into the listening room and direct them at the wall, behind them. He uses this as an extreme illustration of how wrong most speaker evaluation is, particularly by those who say that proper power response is all that is important-because what you will hear is very far from a proper reproduction of the input. We may preserve the power response, but by involving the wall, we lose everything else!

Let us consider the inter relationships between our loudspeaker, the room, and the listener; in light of the psycho-acoustic knowledge we have gained, particularly concerning how we perceive. We recall that the hearing mechanism separates and perceives differently that data which is separated far enough in time from the first data; that, if not separated enough in time, the hearing mechanism smears the data, a "time smear", that.

Long ago the BBC researchers concluded that a monitoring loudspeaker must be free standing, away from walls and especially comers. We know why-if the sound can be made to wrap around the cabinet, and strike the room boundaries at a more distant time interval, the human hearing mechanism will interpret the first arrival from the speaker; and the later arrivals from the room boundaries differently, even to some extent ignoring the later room effects (we will <u>not</u> get into that in this article, though it must be mentioned in passing). Thus the auditioner will perceive a pure signal, a monitoring signal.

Being logical, intelligent, and now informed, we immediately see why any loudspeaker designed to use the walls of the room, or to be used in a comer, smack against a wall, etc. is all wrong-because it will cause the auditioner to "smear" the desired "first arrival" with all the "room sounds" that come from room boundaries. The concept of using room boundaries for augmentation, supplementation, dispersion, or what have you are all frequency concepts-absolutely wrong in terms of what we are after, the accurate reproduction of musical timbres.

How far from the wall is far enough? It depends on the size of the speaker, the larger, the farther away. Some dipoles, a design that we don't favor, recommend distances of twelve feet from the walls (to be used in auditoriums, rather than homes, we surmise!). For the sensible, accurate loudspeakers we recommend, which we are in process of completing, a distance of twelve inches is often enough, though the large monitoring speakers, including those of the BBC, are often better at two feet or so away from the wall.

We still haven't exhausted the subject of the loudspeaker in the room. Let us return to the loudspeaker that is flush against the wall, making the wall signals very close in time to the first arrival signals; thus destroying the time accuracy as perceived by the human hearing mechanism. In addition, this loudspeaker makes the wall an integral part of the reproducing chain, so that everything then tends to sound like the reproducing room, and its peculiar sets of eigentones (resonant structures), merging into the original signal.

As soon as we move any loudspeaker out from the wall, we excite fewer of the eigentones, less intensively; and we delay the onset of that structure, so that the hearing mechanism can now separate the first signal from the reflections from the walls, creating the eigentones; denoting the first signal as the timbre of reproduction, the wall caused eigentones as the "reverberation" or "color" of the reproduction. The hearing mechanism shows a remarkable ability, given the chance, to make this separation; which explains why otherwise impossible listening rooms can often not influence the perceived reproduction as much as the curves of such rooms would indicate!

Let us now consider, however, a method of making a loudspeaker that will be even less room sensitive. Peter Walker, in his original articles on the ESL, showed that a dipole, radiating primarily fore and aft (back), minimized room problems in the bass range, because it did not excite the side walls, the ceiling, and the floor! Let us follow his thought (we should, he is another of the very few geniuses in this art), by asking; can't we minimize even more the interaction of the bass drive system with the front and back walls?

In order to visualize this, and then attempt to solve further this room equation in reproducing music (Peter Walker seems to feel this is the last remaining real problem), we must retrace our steps, to the heyday of air suspension, back in the 1950s. At that time, Edgar Villchur, who had "invented" air suspension (it was later proven to have been an earlier English development, then discarded as not being very accurate!) wrote a long article, showing how important exact placement of his AR speaker was. He showed a multitude of grossly distorted room bass measurements, the response changing rapidly with very small changes in positioning. He said that a prime reason was that most of the bass from his design went around his cabinet and back to room boundaries, so that the room position was all-important, because most of the bass heard by the listener at any one position came from the walls of the room!

He was reciting the classic illustration of a spherical sound source, which he had invented; his critical placement curves showed how violently a spherical, resonant source intermingled with room resonance. Now, does this represent what we have to contend with, with any speaker in any room? Not at all!

First, if we can get rid of spherical propagation in the deep bass, we can improve things. Bailey, in his 1965 articles, mentions an important, though still neglected to this date, design feature. He mentions that two spaced sources of very low frequency data, such as those found in his transmission line, by controlling diffraction effects, thrust the energy forward, rather than around the enclosure. This is the "plane source propagator" theorem of Arthur Radford and later researchers, including the author-such a propagator is by definition less involved in room problems.

Second, if we can substitute for a highly resonant propagator, one that is basically aperiodic; one with much flatter impedance characteristic, which indicates that the amplifier and the voice coil are in much closer liaison; we can predict that the room problems will be minimized, by a virtual short circuiting of them-by placing a damped resistive component across them, rather than a highly resonant reactance. Do you recall our discussion of impedance curves, when we were treating crossovers; when we said evidence is mounting that the impedance curve may represent, more than the earlier experimenters thought, the actual frequency performance under violently shifting conditions?

Please note that the recommended bass enclosures for our ideal or near ideal loudspeaker are either the transmission line or the acoustically damped ducted system. In addition to their other virtues they interact less with the room, so project more of the accurate data to the auditioner than other possible radiators.

Finally we are approaching the conclusion of a long foray into the ever-fascinating (for the author) subject of the loudspeaker, past and present. We must have gained some new insights, particularly if we have read so far and so long. We have established the superiority of the aperiodic loudspeaker for listening to music, if not for sales; we have been able to understand the vital importance of point source and expanding sound source; the relationship of the loudspeaker to the room, and the methods for optimizing this relationship so that the human hearing mechanism will be least upset.

We take our formalized speaker, set it up in a stereo pair, away from walls, and play a record. What can we expect? Well, most anything, because there are just as few records made in approved acoustical fashion (Blumlein, again) as loudspeakers. The typical reaction to the first hearing of an accurate loudspeaker is a slight sense of disappointment, typically, "every record sounds different." Or, “It doesn't have that warm feeling that I like."

The price of superior loudspeaker performance can often be that-less immediate, visceral satisfaction. However, in the words of Raymond Cooke, that loudspeaker is "better" which, over the course of many hundreds of program inputs, will prove to be more satisfactory. The author adds his own, related observation, gained from listening to many thousands of program inputs, and many hundreds of loudspeaker designs over the past fifty years or so-that loudspeaker is more satisfactory which reveals more of the ambient differences which exist from program to program, hall to hall, recording method to recording method.

Ruled out by either conclusion are speakers that have artificial ambience from room back slap, from "air around the instruments"; for these will not have the great variation from program to program about which we speak.

The author also offers another clue to a better loudspeaker; it always seems to increase perceptibly the dynamic range of the program. The author relishes a favorite trick he has used for over twenty years-to play discs from the dawn of stereo to auditioners, who marvel at the progress of the "new" recording methods. Actually, anyone who had read through all the data on linearity, accurate time, attack sounds, and so forth might expect something like this, more accurate reproduction of the dynamics of music; and, since it is the opinion of many that the dynamic range of live performance is the single most important thing that still separates “live” from "reproduced", this is all the better.

As we enter the technologies that promise us not the 20db of dynamic range of the earliest stereo discs; the 30db dynamic range of the first London-Decca "Ring" operas; the 50db of the later analogs; the 60db of the analog from digital tape discs; but 90db, we can' expect, from our near ideal loudspeaker, even more of the intense experience that is the live experience; the reason for having an accurate loudspeaker, when all else is said and done. With the thought, that the accurate loudspeaker is a more enjoyable loudspeaker, we close.

Irving (Bud) M. Fried