1983

Audio Magazine: proposed series Chapter 01 - TDS

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TYPE B:AUDIO.CH1

CHAPTER 1

INTRODUCTION

For the past ten years, AUDIO loudspeaker reviews have been performed using a testing method called time delay spectrometry (TDS). When these tests were first introduced I wrote a brief description of the particular instrumentation which I was using at that time to implement TDS measurements. Because it was such a new concept to most readers and because we were introducing some new measurements to the audio industry (such as time delay corrected phase response and the energy-time curve) it was necessary to give only a cursory description of the method. I believe that it is now time to provide AUDIO readers with a more thorough discussion of the concept, and even of the philosophy that underlies the concept.

Philosophy? You bet. For although most persons might only want to use TDS for simple measurement purposes, much deeper concepts are involved - concepts that relate to fundamental principles of audio measurements.

First, an overall broad-brush picture. Time delay spectrometry is the name I have given to a method of test and analysis which involves the implementation of a new class of integral transform. TDS derives its unique testing capability from the mathematical properties of this new transform. So that there is no misunderstanding, let me point out that the conventional Fourier integral transform is only a special case of this more general transform. The Fourier transform is a degenerate form of this new transform. In an active stimulus-response situation, such as acoustic testing of rooms or loudspeakers, any Fourier transform implemented measurement can be duplicated by TDS, but the converse is not always possible.

The new integral transform, in turn, derives from an approach to the meaning of "measurement" or "observation"; an approach expressed in what I have referred to as a Principle of Alternatives. As if this were not abstract enough, the entire package can be considered to be the expression of a new paradigm of analysis, and one result of this paradigm is a set of demonstratable laws governing the partitions of energy density in the measurement of a linear system.

So what has this to do with audio, and in particular to loudspeaker measurements? Plenty, because I was led to these considerations through my interest in trying to understand how to reconcile objective measurements with subjective perception. In my attempt to try and understand why "what we measure" and "what we hear" do not always agree, I was led to an entirely different approach to the mathematical structure that might underly such things.

What I plan to do over the next few months, with the kind permission of our editor Gene Pitts, is present a series of articles which outline this different structure of analysis. These articles will be conversational in nature, somewhat like an informal discussion. Each article will build on the material which was presented in the preceding articles. I will be assembling the equivalent of a small course that covers the fundamental aspects of this new approach to analysis, an approach that is not covered elsewhere, either in books or classroom. Where I feel it is necessary for a proper understanding of the material, I will not hold back on the mathematics. This is done for accuracy, not to snow anybody. The math is never as important as the spirit of this new approach so I will, where possible, try to say in words what the math symbolizes. The discussions will range from philosophy through abstract math to "how to" methods of audio test, with special emphasis on the proper use of TDS.

The reason for such diversity is quite simple; we're playing a brand new ball game. Just how new this ball game is can be appreciated by a brief discussion of the new integral transform which I referred to earlier.

AN INTEGRAL TRANSFORM

In contemporary analysis, the class of integral transform having the form shown in relation (1), where the "kernel" has the specific symmetry properties shown in relation (2), is called the generalized Fourier transform. The integral is taken with respect to Lebesgue measure and there are a few additional mathematical considerations on the nature of "distribution" over which this transform is defined and on the rate of increase of the distribution at infinity. Right now, I won't go into these niceties, since they are not important when dealing with the class of signals which we presently handle in audio.

In conventional signal theory we are familiar with the type of Fourier transform in which the kernel is composed of a complex exponential whose angle is represented by an inner product. The kernel is given in relation (3) and the inner product is given in relation (4). This integral then takes the familiar shape given in relation (5), where the constant K is chosen such that both f(x) and g(y) have equal sum squared magnitude, a condition first stated by Rayleigh and later called Plancherel's Theorem. Relation (5) is the "Fourier integral" which is normally referred to in signal theory.

In simple audio form, the one-dimensional Fourier transform between a time signal, call it f(t), and a frequency signal, call it F(omega), looks like relationship (6). The inverse Fourier transform, which expresses time in terms of frequency, is shown in relation (7).

Don't get hung up with where I put the 2pi. Where you put it isn't important, as long as you are consistent in the way you handle the computations.

These simple little relations, (6) and (7), are virtually the sum and substance of what we euphemistically call the "frequency domain" and "time domain". You cannot imagine the large number of books, papers and articles which have been, and are being, written around these relations. They beat these simple relations all over the block; and just about the time you think nothing more can be said about the subject, someone else comes along with another mallet and whacks away at some other aspect of the equations.

Sure, they're important, but, good grief, are they that important. And in particular, are they the end of the analysis road?

No, they're not the end of the road; they're not even a major highway. In the coming articles I will go into greater details, but for right now let me go back to the beginning and define a general class of map under the relation shown in (8) where the mapping kernel is no longer symmetric in y and x, but can still be expressed as a complex exponential whose angle, this time, is represented by a hypersurface in y and x. The new mapping kernel is given in relation (9).

Take a look at relation (4), which is used in the Fourier transform. Relation (4), called an inner product, is also the equation of a hyperplane in coordinates x. It is the equation of a perfectly "flat" surface in N-dimensions. In three dimensions, this is a plane; in two dimensions it is a straight line; and in one dimension it is a point.

In three dimensional geometry, like the one you and I are accustomed to thinking about when we talk about physical objects at a moment in time, a flat piece of paper is a part of a plane surface. Think of a flat piece of paper as representing this plane. Now imagine crumpling the paper. What is the equation of this crumpled piece of paper? It's not relation (4); that's a flat piece of paper. There must be some other equation which represents a crumpled piece of paper. And more importantly, whatever the equation of this new crumpled surface is, it must include the flat piece of paper as a special case.

A general term for all possible pieces of paper, crumpled as well as flat, is the word hypersurface. The hypersurface shown in relation (10) obviously includes the "flat" hypersurface given in relation (4). The mapping kernel of this new integral transform uses this hypersurface as a complex angle, relation (9).

The new integral transform has the form shown in relation (11) with the "inverse" transform having the form shown in relation (12). Obviously, (5) is a special case of (11); and (5) is to (11) as the geometry of a flat piece of paper is to the geometry of all possible pieces of paper, crumpled as well as flat. TDS is the implementation of relations (11) and (12). Contemporary Fourier analysis is the implementation of relations (5).

If you are into math analysis, you are probably experiencing one, or both, of two reactions. The first reaction is: what absolutely audacious right do I have to claim that the Fourier transform is a special case of something I have developed. The second reaction is a chilly feeling: what if I am correct.

If you are not into math, you are probably experiencing another more painful type of reaction. Namely, what the heck am I doing in presenting these math chicken tracks in a magazine devoted to audio?

I'll get around to explaining the math behind relations (11) and (12) in future articles (And, quite frankly, I stuffed these equations right up front in these discussions in order to get your attention that I am going to be discussing something that is really new, not some warmed over toast). But right now, I believe that the "non math" reader deserves an explanation.

AN AUDIOPHILE'S SEARCH

I will now present a little background to this matter, and a story which I often tell about myself. I have been an audio hobbyist since my high school days. I maintained this interest through college and into "work", and I managed to win a number of patents on early transistor audio amplifier circuits. I was aware, as were others, that what I heard did not always agree completely with what I measured, using the techniques of the day. Armed with a reasonably good math background, I assumed that I could overwhelm the situation with mathematics and understand what was going on. How wrong I was.

After many frustrating years it finally occurred to me that although the mathematical structures with which I was familiar were correct, they may not have been applied to the proper problem-solving model. To use a concept proposed by Thomas Kuhn, the paradigm was inappropriate to the tools I was using. So I set the tools aside and took a harder look at the model.

Although my intellectual trajectory was somewhat similar to that described by the ball in a pinball machine, was led to the following conclusions. First, the words and phrases which we use for subjective descriptions, no matter how flowery they may be or unintelligible to the purely technical person, nonetheless constitute a language capable of conveying meaning at the experiential level. Second, there is strong cross-sensory association in this language, such that the description of an evoked experience due to sound may contain visual, tactile and other sensory associations. This seemed true for single as well as multiple sensory stimulation. For example, what we saw could influence what we heard.

There were a number of other observations along the way, but the most significant, to me, occurred once I recognized that the descriptive terminology constituted a rudimentary language. The technical details are discussed in the Journal of the Audio Engineering Society, but in effect what I did was to write each term of subjective audio, that I could find, on a separate piece of paper. Then I imagined that all these scraps of paper were scattered on a table top. One by one, I mentally began to assemble the scraps into piles such that each pile seemed to represent something that I might be able to describe in a technical (that is, objective) manner. For example, a "bright" sound and a "warm" sound seemed to describe a tonal attribute; whereas a "smeary" sound seemed to describe a spatial attribute. You may not agree with my particular categorization, and that is all right, but the one inescapable fact that emerged was that I was unable to combine the scraps into less than five piles.

And then the light dawned, because each pile, in effect, represented a different dimensional attribute. This, I felt, was a clue why subjective and objective audio could not talk to each other; they existed at different levels of dimensionality.

This was a numbing thought to me. We had been performing one-dimensional analysis (for example, volts as a function of time) on an audio signal. If subjective perception were, at least, five dimensional, then this could explain why the subjective and objective person could not understand each other - yet both could be correct in their own frame of reference.

Consider the following analogy: If you hold up a coherent hologram of a three-dimensional object and view it in normal incoherent light, you cannot see the three-dimensional image. All you see is a two-dimensional pattern that looks like a fight between two screen doors. Such a hologram is a three-to-two map. It is a map of a three-dimensional scene onto a two-dimensional format. Now imagine how you might have felt if, never having heard of a hologram, someone handed you a two-dimensional piece of film and said, "Here's a picture of my family, what do you think of them?". You couldn't make heads or tails of it. Then you might pull out your favorite family picture and hand it to that person, who, viewing it under coherent laser light might reply, "I don't see anything". You might be tempted to engage that person in a heated debate concerning his vision, as well as his sanity. The only way you could "see" his image would be when he provided a two-to three map that, in effect, allowed you to transform to his frame of reference.

Perhaps this foregoing scenario is a poor analogy, but you can appreciate why the concept of frame of reference, and of the dimensionality of that frame of reference, might be important in the communication of ideas between subjective and objective audio. I found this concept fascinating. Of deeper impact, to me, was the recognition that I was dealing with a structure of analysis in which the same thing could, somehow, be represented by completely alternative descriptions at different levels of dimensionality.

Dimension is a topological invariant; hence I could not depend upon using the common mathematical tools of topology. This was devastating. It took away all my tools. I would have to start from scratch if I wanted to come up with a viable math structure.

Furthermore, this revelation (that I could not depend upon topology) led to a most interesting concept. A topological space (to use the math terms) is a set from which has been swept away all structure irrelevant to the continuity of functions defined on it. There may not even be a metric. In other words, the closeness of one attribute to another may not necessarily be expressible as a "distance". It is entirely possible, for example, that we could not develop a "distance ranking" of how good or how bad one audio reproduction is relative to another when both are presented on the same frame of reference.

Such things did indeed lead to a different mathematical structure of analysis. And several new tools, including TDS, have resulted from this new structure. It is this new structure, and one of those tools, TDS, which I intend to outline in the forthcoming series of articles.

At first glance, afficienados of subjective audio might probably wonder what all this seemingly technical material has to do with "how it sounds". It has a lot to do with "how it sounds", and AUDIO is perhaps the best place to present this material, because all of this is a direct consequence of trying to find some objective way of dealing with subjective perception.

If you happen to be a technologist, don't turn away just yet, because, to put it in the venacular, "a funny thing happened on the way to trying to understand how it sounds". You'll find that this new structure provides some most interesting insights into some very old unsolved problems in science.

In the next article I will discuss the new paradigm.