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Unpublished Writings from Audio Magazine: proposed series Chapter 1 and 2

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Chapter 1

INTRODUCTION

For the past ten years, AUDIO [1] 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. [2]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 underlie 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

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[1] Heyser was a contributor for Audio Magazine for many years. Audio Magazine was a periodical published form 1947 to 2000.

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),

$$g(y) = \int_x K\,(y,x) f(x) dx \qquad (1)$$

where the "kernel" has the specific symmetry properties shown in relation (2),

$$K(y,x) = K\,(x,y) \qquad (2)$$

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)

$$K(y,x) = e^{i(y,x)} \qquad (3)$$

and the inner product is given in relation (4).

$$(y,x) = y_1 x_1 + y_2 x_2 + \cdots + y_n x_x \qquad (4)$$

This integral then takes the familiar shape given in relation (5),

$$g(y) = \frac{1}{K} \int_x e^{i(y,x)} f(x) dx \qquad (5)$$

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(τ), and a frequency signal, call it F(ω), looks like relationship (6).

$$F(\omega) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} e^{i\omega\tau} f(\tau) d\tau \qquad (6)$$

The inverse Fourier transform, which expresses time in terms of frequency, is shown in relation (7).

$$f(\tau) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} e^{-i\omega\tau}\ f(\omega) d\omega \qquad (7)$$

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).

$$g(y) = \int_x m\ (y, x) f(x) \qquad (8)$$

$$m(y, x) = e^{i\phi(y,x)} \qquad (9)$$

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.

$$\phi = (y, x) \qquad (10)$$

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).

$$g(x) = \frac{1}{K}\int_x e^{i\phi(y,x)}\ f(x)dx \qquad (11)$$

$$f(x) = \frac{1}{K}\int_y e^{-i\phi(y,x)}\ g(y)dy \qquad (12)$$

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, aficionados 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 vernacular, "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.

CHAPTER 2

INTRODUCTION

For a long time I have been concerned with the fact that what we "measure" does not always correlate with what we "hear". Rather than sit back and blame it on the vagaries of subjective perception (after all, isn't the math always correct?) I have been engaged in trying to find a better math structure that might be capable of dealing with subjective perception as well as including our present math structure as a special case. As wild as that sounds, I believe I have gone a long way toward reaching this goal. It is not an overnight wonder; I have been working on this problem for nearly twenty years. TDS is one consequence of this work. In these brief articles I am presenting the technical basis of that newer math structure.

In the last article I jumped in with both feet and introduced a new integral transform and showed that the Fourier transform was nothing more than a degenerate case of this transform. The new transform is the basis for TDS.

This new transform can be regarded as a key, which we can use to open doors to domains of analysis far beyond the traditional time domain and frequency domain of contemporary theory. But, unless you know how to interpret this key, you might never know how to use it, let alone know that such doors existed. In that first article I handed you the key, with the intention of demonstrating that we are dealing with something completely new, and not some warmed over version of contemporary analysis. In this, and in the upcoming series of articles, we will learn where that key came from, how it can be used, and whether other keys might exist. Now let's go down to the most fundamental level and begin our journey by considering a new paradigm.

A PARADIGM IS NOT FOUR NICKELS

I use the term *paradigm* in the sense proposed by Thomas Kuhn. In a problem solving situation, we generally resort to some conceptual model. This conceptual model is the way we "think about" a matter when trying to explain that matter to ourselves, or to others. The term paradigm has been variously applied to this situation. Paradigm may refer to the particular problem solving model itself, or in some cases may refer to a collection of persons all of whom use the same problem solving model and who, taken collectively, represent a particular discipline of thought. I will use paradigm in the sense of the model itself.

It is my contention that the mathematical structure which we use for signal analysis has a paradigm at its heart. This paradigm is never discussed and is so much a part of the way we set up problems that we might never consider it to be a paradigm until it is pointed out as such.

This hidden paradigm is revealed in the procedures that we use in traditional analysis, for we always try to describe something that is complicated in terms of things that are simpler and that are expressed in the same type of terms as the more complicated thing we're trying to describe. Therein lies the paradigm, for what we are assuming is that there are other simpler signals, *expressed in the same frame of reference*, which when properly combined, can duplicate the messy signal.

This procedure is best exemplified in audio analysis by the traditional Fourier sine and cosine series approach to what we call "harmonic analysis". If an audio signal keeps repeating its form over and over, such as a square wave, then we know that the time waveform of this repeating signal can be duplicated by adding a series of time waveforms which are made up of sine and cosine shapes of appropriate size. There is absolutely nothing wrong in this approach, and its mathematical pedigree is beyond reproach; but, without realizing it, this hidden paradigm, that it takes time waveforms to synthesize a time waveform has locked us into a "way of thinking".

PRINCIPLE OF ALTERNATIVES[3]

Having recognized this as a paradigm, let me now suggest that we modify the paradigm in such a way as to expand the importance of "frame of reference", yet include everything which we now use in problem solving.

Suppose I presume, as my paradigm, that nature doesn't care how we choose to look at her; there is no preferred frame of reference for representing nature.

Now, what do you mean when I say there is no "preferred" frame of reference? I really mean that I assume there are a great many different ways of describing nature, that each of these ways is totally complete in its capability of characterizing nature and doesn't depend upon the existence of any other way, and that there is nothing accounted for in one way that cannot be accounted for in any of the other ways. What I am assuming is the existence of a basic principle, which I shall call a principle of alternatives. Each of the different ways of "looking" at nature is an alternative which we may use.

The name I will give to each of the independent ways of characterization is "alternative". There are all sorts of alternatives. If we categorize a bunch of alternatives under some agree upon set of conditions, call those conditions C, then that group of alternatives will be said to be "alternatives under conditions C" or, in shorter form, "C-alternatives".

We ought to be able to find some map which can take us from one C-alternative to another C-alternative. In other words, if you "see" nature in terms of some particular frame of reference, and if I "see" nature in terms of some alternative frame of reference, and if there is some agreed upon condition that we might call a "law of nature" (that cannot depend upon any particular coordinate value), then there must be some way of converting my view of nature to your view of nature. There should be some map, m, to convert my view, f, to your view, g, under conditions C. In math symbolism,

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[3] This concept of the principle of alternatives is more fully explored in *Alternatives*, (Audio volume 62, no. 2 pp. 50-52 1977, June)also reprinted in the *Time Delay Spectrometry, An anthology of the works of Richard C. Heyser,* AES publisher, page 144. A concise summary is also included in the posthumous manuscript " Fundamental Principles and Some Applications of Time Delay Spectrometry" page 212 in the AES Anthology.

$$m: f \xrightarrow{C} g$$

This can be interpreted in audio terms. The name of the game in audio measurements ought to be: how can I measure (f) what I hear (g) and find some way (m) to explain it. Or, more to the point, if the person who measures (f) never speaks the same language (m) as the person who hears (g), then audio measurements might as well be thrown out of the window.

CONDITION C

What about the conditions which I have labeled C? One of the things we strive for, when trying to come up with what we believe are fundamental relations, is a way of deriving as many subsequent relations as we can from this fundamental set. One has a gut feel that we're not dealing with fundamental laws when we have to tack on things which we can observe and call then "laws of nature". The more "laws of nature" we have to assume in order to make things work, the less we really know about what is going on. Let's see what we can pull out of this principle of alternatives. If there are an unlimited number of alternative ways of "looking" at nature, then it is reasonable to ask whether there is any property that does not depend upon "way of looking". Is there some fundamental property that truly characterizes an event and is not changed when we move from one alternative to another? In math parlance, what is invariant under map m?

Nothing that depends upon "frame of reference" can be depended upon to remain invariant when we map among alternatives. Connectedness and continuity cannot be depended upon, nor can the number of dimensions. One by one, the nice comfortable properties of description fall as we look at more and more alternatives.

But there is one, and as far as I have been able to determine, only one, important property that stays the same - alternative to alternative - when we consider the important descriptions of nature that we may want to use. That property is the net "how much".

To see this, consider that we have developed a legitimate description, call it F, of a natural process in terms of a frame of reference having coordinates x. This is then some F as a function of x, call it F(x). Some other being, perhaps one who utilizes an eight dimensional frame of reference, also has a legitimate description. If his (or its) frame of reference has coordinates y, then that is some G(y). And yet, another being, with coordinates z has an H(z). If F(x), G(y) and H(z) all represent alternative descriptions of each other, then what process removes all x, y and z dependence from the descriptions and results in something that cannot depend upon anyone's coordinates?

A procedure that does this is one that reduces each description to the same scalar. It reduces the dimensionality to zero. It is integration. But we cannot just use our ordinary integral. We must use one that can handle really wild and pathological (by our conventional standards) descriptions. The integral should be taken with respect to Lebesgue measure. This means that relation (13) must hold, where E is a number.

$$\int_x F\,(x)dx = \int_y G(y)dy = \int_z H(z)dz = E \qquad (13)$$

ENERGY

There is a scalar entity that is conserved when all aspects of a description are considered over the complete frame of reference. Sound familiar? Sure, it's what we call energy. But rather than postulate the conservation of energy as the fundamental law, we've come up with something rather startling. As a consequence of the principle of alternatives, there must be a property that is conserved, and this property can be called energy. We derived it from the principle.

Suppose we have established a description in a valid frame of reference which uses coordinates s. We have some E(s). Since the net Lebesgue sum is a constant E we know that relation (14) must hold.

$$\int_s E(s)ds = E \quad (14)$$

In more conventional terminology, E(s) is an expression of energy density as a function of coordinates s, whose net sum over all s must be a scalar which we call total energy, E. As it stands, this relationship is so general as to be of little utility. However, in searching for a more useful tool to use in my paradigm, I came across a relatively obscure math theorem, having nothing to do with energy, that presented an answer. If I define a new entity, call it h(s), the square of whose magnitude is equal to E(s), then I can cast relation (14) into a manageable form. Defining h(s) as the complex entity given in relation (15), we have the expression shown in relation (16).

$$h(s) = f(s) + ig(s) \quad (15)$$

$$\int_s E(s)ds = \int_s |h(s)|^2 \, ds = E \quad (16)$$

Bingo. Since E is finite (that is, E is a number less than infinity), we have a relationship that states h(s) is square integrable, and consequently will be of a class known in mathematics as L^2. The L stands for Lebesgue and the superscript indicates squaring. I use the superscript notation. The subscript notation L_2 is often seen to identify this class of integral. The condition C now becomes Conservation of net Lebesgue complex square summability.

Although not apparent at this point in our discussion, the expression of relation (15) is exceedingly important. I call h(s) the energy functional of coordinates s. h(s) contains all the energy density relationships expressed in terms of coordinates s. Things that fall out of this energy functional have other names. One of the things that falls directly out of the energy functional when dealing with the time domain properties of systems, such as loudspeakers, is what I have called the energy-time curve (ETC) and which we've presented in all of our loudspeaker reviews.

The math theorem that I uncovered is due to E.C.Titchmarsh. It is rather startling. It states that the strongest of all possible math conditions demand that the f(s) and g(s) parts of any h(s) MUST be related in a very special manner. The necessary and sufficient conditions that h(s) be square integrable is that f(s) and g(s) be related by a special process known as Hilbert transform. If the frame of reference is eight dimensional, then there will be eight s coordinates and eight corresponding h's, each with its own f and g parts. Each of the f parts will be related to its corresponding g part by the Hilbert transform.

Discovering this fact was something akin to turning the light on in an otherwise dark room. It let me go from something that was so general as to be useless, to a full set of mathematics that predicted what kinds of things we should expect to find in nature when using this new paradigm.

A MYSTERY

In the next article we will explore these new energy relationships. Until then you might find it amusing to contemplate the circuit shown in figure 1 and try to unravel the mystery that might be called "The Case of the Missing Energy".

On Monday, an advertisement is printed in the local newspaper announcing a contest in which a prize will be awarded to the first person who can transfer energy from a battery to a condenser, under linear passive conditions, without losing half or more of the transferred energy. For a mere entrance fee of one dollar, the lucky winner will receive a cool thousand dollars. The contest closes on Friday.

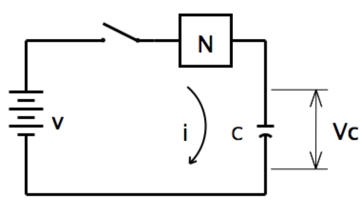

Figure 1

The experimenter is supplied with the network shown in figure 1. The capacitor in connected through a black box, shown as network N, to the switch that will allow current to flow from the battery when the judge starts the contest. The experimenter is not allowed to know what is in the box, but is told that, whatever it is, it will not have any net voltage drop when the charging current finally drops to zero, and that the box does have a dc continuity, so that the capacitor will eventually be charged to the battery voltage. The rules are simple, the experimenter must put his own network in series with N, and his own network may be anything of his choosing, so long as it contains only linear passive components, such as resistors, inductors and capacitors, and so long as it is truly in series with N and has no third terminal connected to common. The judge will start the experiment by closing the switch and will very precisely measure the exact amount of energy drawn from the battery. The switch will remain closed until the current stops, or until the current is so small that even the judge can no longer measure any energy being drained from the battery over a period of time. Then the switch will be opened and the net amount of energy in the capacitor will be measured.

One by one, various experimenters try their hand; only to discover that the best they can achieve is a loss of half the energy delivered from the battery. Somehow, half the energy always gets lost in the box, no matter how hard they try.

Our villain, Snavely vonPanhandle, breaks into the testing laboratory before his own turn and discovers that the network contains a light bulb, a resonance circuit tuned to Channel 2 complete with an electromagnetic antenna that can send the energy off to Afghanistan, and an electric motor that drives a pump which lifts water up a standpipe while current flows from the battery, and which reverts to a generator when the water drains back down and the charging current drops off. No fool, Snavely reasons that the judges have set a trap for the unwary, not to mention the innocent. So Snavely manages to appear at the laboratory before the judge arrives, and craftily drills a hole through the box, connecting a short circuit that bypasses all these energy consuming parts.

The day before, the previous contestant had managed to get half the energy from the battery into the condenser by simply letting the network charge the capacitor directly. Sure of his victory, Snavely smugly opted for the same setup, knowing that he had short circuited all these energy consuming internal items. But, so as not to raise the judge's suspicion, Snavely placed a one millihenry inductor in series with his capacitor, knowing that the inductor had no internal resistance which could dissipate energy.

Imagine Snavely's surprise when the judge announced that he, too, had only managed to get half the energy from the battery into the capacitor. Stunned, Snavely grasped the millihenry inductor. It was ice cold. Where had the energy gone?