"TECHNICIANS' DAY"
THURSDAY MORNING - JULY 22, 1965

MR. ARCHER TAYLOR: We will open our Technician's Day Session. For several years NCTA has had a committee studying various aspects of Industry technical standards. This has proven to be a very difficult and very important job. It's, therefore, quite gratifying to me that this demonstration which you are about to see this morning could be produced under the sponsorship of the Standards Committee of NCTA. We have in our industry both in the manufacturer's design laboratories in the operation of systems a remarkable degree of technical confidence. A very large number of these engineers and technicians have participated in putting together this project to to demonstrate to you this morning.

I would like to present to you to tell you about the demonstration Mr. H. J. Schlafly. Mr. Schlafly is Senior Vice President of the Teleprompter Corporation, one of the multiple owners of CATV systems. He is formerly President of the Telepro Industries, Incorporated, served for six years as Director of Television Research for 30th Century Fox, was in the advance development laboratory of General Electric Company. He holds a BSEE at the University of Notre Dame and has done graduate work at Syracuse University. He is a Fellow of the Society of Motion Picture and Television Engineers and is a Senior Member of the Institute of Electronic and Electrical Engineers. Mr. Schlafly. (Applause)

MR. HUBERT SCHLAFLY: Thank you, Archer. The subject to this session is Measurement of Noise and Cross Modulation. I am representing the NCTA Ad Hoc Committee on Technical Standards and am appearing here in no other capacity.

Now, noise and cross modulation are at least two of the items in the life of a CATV operator that he could just as well do without. Most of us are familiar with noise. This is snow, dancing dots, picture grain or whatever you wish to call it. It spreads over the entire picture area, provides distraction to the viewer, then loss of picture detail and finally it curtains off the entire view.

Noise is random, it has no pattern, shape, form or period. It is unwanted signal, containing no intelligence which competes with the picture. We know a good deal about random noise (some call it thermal noise). We can see it, we understand it, we know how to measure it. I think that due to the shortness of time, with your permission, I will jump over any further remarks on random noise, and go into ar equally serious area of concern to us - unwanted signals.

This kind of picture interference, I call the "not-noise", or the "not-interference." It is <u>not</u> random, it is <u>not</u> easily understood, it is <u>not</u> easy to measure and after it is measured, the measurement is <u>not</u> easily interpreted. This is cross modulation.

Cross modulation is an unwanted byproduct of any amplifier that is less than perfect as a linear amplifying device. If the output signal at any operating level is not a constant multiplying factor of the input signal or is not a constant number of dbs added to the input level (for those of you who like to think in dbs), then, like the "Music Man" with 101 Trombones, "brother, you've got trouble in Central City".

Now, in order not to confuse us oldtimes who understand tubes but aren't so sure we understand transistors, let's refer to an amplifier as having a transfer characteristic. If this transfer characteristic is linear, everything is great. If it is nonlinear over the range that the designer choses to operate, you've got a whole beehive of possible signals that you never knew existed. If the transfer characteristic changes its slope over the used range, you have second order effects. You have second harmonics and second order beats. Thanks to luck and to the foresight of the original system engineers and to the understanding of the engineering staff of the Federal Communications Commission, VHF channels were selected so that second order effects do not bother us. They exist, but they don't fall into the frequency bands in which we work.

But, the rate at which the transfer characteristic changes slope produces third order effects and these do bother us. Now, amplifier gain and level enters into this picture, because although an equipment designer can find some portions of a transfer characteristic that is linear, the greater a portion of the transfer characteristic must be used the greater is the chance for nonlinearity. So the harder you drive an amplifier, the less linear it is likely to be and the greater will be these unwanted internally generated signals which we call cross modulation and intermodulation.

Low level amplifiers generally have good linearity and have very low levels of these unwanted modulation effects. High output level amplifiers, let's say in the range of 40 to 50 DBMV can have increasingly disturbing levels of this modulation. And, it increases rapidly with output level. A fairly accurate rule of thumb for a typical amplifier is that every 1 db of increase in output level produces about 2 db increase in the relative cross modulation interference.

Third order nonlinearity generates beats (beat frequencies or spurious signals) that do fall in portions of the spectrum that we use. These generated frequencies are referred to as:

$$\text{Intermodulations} = 2f_{a} \pm f_{b}$$

$$\text{Third Harmonies} = 3f_{a}$$

$$\text{and Triple Beats} = f_{a} \pm f_{b} \pm f_{c}$$

My analysis (which I don't guarantee to be without error) shows that only two video carriers, channels 3 and 4, have Third Harmonies which fall in an upper band channel.

Intermodulation products of the video carriers are confined to a reasonable number of frequencies in the TV channels (over half of which are avoided if adjacent channels are not used). Triple beats, which can occur almost anywhere, have not been analyzed by me on a statistical base.

Since the peak of sync has the greatest amplitude in TV signal, this unwanted vertical bar is noticed first. Of course, it's not that the video modulation of the unwanted channel isn't there, it's just that it's at a lower level and you don't see it. If the synchornizing frequencies of the two television signals on different channels are stable and constant, this vertical bar stands still in the picture. This may, in some cases, be less annoying than if they were not synchronous, in which case they do not stand still and you get the familiar windshield wiper effect. Cross modulation is hard to measure and since this annoyance is largely subjective, just as beauty is in the eye of the beholder, it is hard to evaluate. It is most easily seen on the blank screen. From here on we're going to work with an actual demonstration showing both broadcast picture and a closed circuit camera pick up of the oscilloscope display on this 8-foot wide screen by means of this TV projector.

There is a setup of equipment in the back room, intentionally in the back room, because we want to make no reference in this session to any particular philosophy or any particular manufacturer's methods or equipments. We have twelve picture input channels and we have a CW generator which can replace any one of the 12 pictures. All these signals are combined and fed to an input attenuator.

The output of that attenuator goes into the amplifier that's under test. The output of the amplifier goes to an output attenuator which cuts down the combined signal to a suitable level to put into a television receiver. In this case, the television receiver will be a TV projector and we'll put in on the large screen. As the input attenuation is decreased, meaning a higher drive signal, the amplifier will operate over a larger portion of its transfer characteristic and will show an increase in cross modulation effect. We adjust this output attenuator so that the level to the receiver is constant.

We also have a setup in the back room so that we can measure output level, not only the level of the CW carrier but also the AC modulation of that carrier, which in this case can only be cross modulation and we can view this on an oscilloscope.

This is a large screen view of the oscilloscope showing cross modulation as it is being measured right now in our back room set-up. Can someone tell me how many channels are on this picture? One? One channel. One interferring channel. Yes. This is one channel which is interferring with the CW carrier. The spike is the peak of the sync pulse. The signal in between those two spikes that you see is incidental video, but the item that's going to cause the trouble is that spike. Now, let's add an additional channel. A second channel of TV picture will be added to the amplifier at the input. Since the sync generators on the stations that we are using are non-synchronous, you will note that the additional interfering channel will drift through. If the picture that you add is synchronous, that spike will increase in size rather than give the drift thru effect. Now I will turn this projector over to the TV picture. That's cross modulation right there.

Although my raster isn't synced since we're running on CW signal rather than on a normal television picture, you can see those vertical bars. Let me turn over to another channel and get this thing tuned up a little bit. Remove one of the interfering channels. All right. Now, its stabilized somewhat.

Of course you are seeing cross modulation on a blank screen where it is much easier to see. If we put this onto a channel which had picture content you might not see it to this extent. But what you do see here is the interference that causes the trouble.

Now change the level of the amplifier. Reduce the output level of the amplifier, remove some of the output attenuation to bring the level back and see if we can observe the reduction in the cross modulation. This is 3 db less than you saw before and you can see that the interference signal (I've lost sync again, lost stability, there. That's better). is substantially less than you saw before and it would be less subjectively annoying on the picture.

Do you want to see a third interfering channel added? Put all of them on now and increase the output level so that we have an easier time seeing it. By increasing the output level of the amplifier we drive the amplifier harder. We use a greater portion of the transfer characteristic and we're getting into the change of gain and to the rate of change of gain which generates second and third harmonic effects. You're getting some beat here also.

We are intentionally overdriving this amplifier by a substantial amount just to accentuate the effect that you see here. Now I'll turn over to a picture channel. You can see the cross mod in the picture itself. Fairly stable here rather than the windshield wiper that many see as more objectionable. It's an amazing thing that the modulation of one picture can sit right on the other picture and modulate that carrier just as though it was an original part of its own signal.

Now, this isn't new. This isn't a new thing that the cable television people dreamed up. In fact, I have a textbook that I used in college (and that's a long time ago) on the table there, "Radio Engineering" by Terman. This book has 5 or 6 pages that give an excellent analysis of the problem of cross modulation.

Now, reduce the output level of the amplifier, gentlemen, by first 3 db and then 6 db and let's watch this reduction. Let's go 3 more. And, 3 more.

I want to remind you, again, that we have been intentionally overdriving this amplifier by terrific amount in order to accentuate the problem here. Bring it down to an acceptable level. Let me emphasize that word acceptable. Since it is a subjective problem, the value can differ. I must admit that we have not as yet agreed on exactly the proper means of test nor on a figure of db cross modulation must be down to be acceptable to the industry. That is the problem on which the NCTA Standards Committee is working.

Now, we have reduced the level of the output of the amplifier. We have removed a compensating amount of attenuation from the output line to the receiver and we have reduced the cross modulation by a substantial amount.

Let's go back to a CW picture showing cross mod and the drift through of sync signals. Those signals look synchronous to me there. Maybe they're flipping through so fast that I don't see them. Yes, now they do add. When they do come together, when they do become synchronous, those spikes add in height and of course you get a much worse condition temporarily. I haven't satisfied myself whether the larger signal is worse than the low frequency flip through.

In order to take a photograph of an additive cross modulation condition you may have to wait ten minutes to see it on the scope and when you did see it, it might last only a fraction of a second.

I just want to say that this demonstration has been a difficult one to put on. It has taken much equipment and has taken much time and thought. This was freely donated by several of the manufacturers who have not asked for, expect, nor will they get a plug, on this NCTA sponsored demonstration. I wish to compliment them and the industry. I have sat in many committees and I have fought for and seen fights for many standards. I have never been in an industry where I have seen as such a wonderful climate of cooperation or spirit of fighting for our common cause against the common enemy as I have in the NCTA. With the recognition that we must make our mark in a highly professional field that includes broadcasters, networks, telephone companies and others, this spirit of cooperation is vital. I do not see how our industry can help but flourish to new growth and success. Thank you. (Applause)

MR. TAYLOR: I would like to point out that we have in this room manufacturers, engineers and system engineers who have done a lot of work in this field. And, in addition to questions, if one of you has a contribution you'd like to make to this general area of discussion, we'd be delighted to spend a little time listening to it. There is a microphone there or right here. Yes, sir. Will you identify yourself?

MR. DICK PECK, WREX TV, Rockford, Illinois: The demonstration that you showed - presumedly the normal amplification of the amplifier itself was somewhere in the neighborhood of 30 to 40 db?

MR. SCHLAFLY: I would say that it's probably in that area.

MR. PECK: How far about that did you drive it?

MR. SCHLAFLY: Well, in this particular case, let's consider the rated output of the amplifier as 0 db and let me get an answer for you from the backroom here as to how far they went. I think a considerable amount. The answer is twelve to fifteen db above normal rated output. And, you recall a rule of thumb was that every increase of db and output level brings about 2 db increase in cross modulation. You know that we were running pretty high here.

MR. TOM MORRISSEY, Denver: I was just wondering if this was an all band or a split band amplifier and you might want to make a comment as to the effect that might have on it.

MR. SCHLAFLY: This is an all band amplifier. The number of channels that you use gives you that many more opportunities for cross modulation. If you have separation of the bands, you can substantially reduce some of the beat frequency or spurious signal annoyances.

In cross modulation, I'm under the impression, Tom, that it's just a matter of how many channels you're using, not necessarily whether they are high band or low band. Would you like to correct that?

MR. MORRISSEY: Well, if we have a true split band amplifier, of course, we have two separate channels, the signals are going through, and this is all I was suggesting, is that there are some of those in use.

MR. SCHLAFLY: Yes.

MR. MORRISSEY: Where the signals simply don't mix. There is no common modulation.

MR. SCHLAFLY: Correct.

MR. EARL QUALM, Long Island Cable Division: I notice that the interference with the vertical bar. Why was it colored white rather than black? Wasn't this a vertical sync bar? And, wasn't it in the black area?

MR. SCHLAFLY: This was a vertical sync bar. You notice that what video information you saw was negative. I think that in the mathematics analysis you come across that inversion of the cross mod. Is that correct, Ken? There is an inversion of sign in this modulation as against the normal modulation on the channel that it is influencing.

MR. JACK THREADGILD, Brady, Texas: You made measurements here with both the scope method and the TV receiver method and from the pictures you showed us, it looked like you could pick out your cross modulation better using raster on a TV set. Did you find this in all your experiments? In other words, which way can you detect it?

MR. SCHLAFLY: The most critical method is the white screen method, but you don't get numbers out of the white screen method. And I didn't metnion it, but the scope there was calibrated in percent of modulation, cross modulation interference, which could be readily read off if we had chosen to do that.

MR. THREADGILD: You can get a value, but the other one will tell you -- it will show up the problem quicker.

MR. SCHLAFLY: This is not an easy thing to measure and I think a good deal of thought is going to have to be given before we come up with an easy way of doing it or with a practical way of ding it in the field. Of course, the end result is the picture itself. And, this varies greater from viewer to viewer. It's like judging hi-fi. If it's your system, you think it's pretty good. (Laughter)

MR. TAYLOR: I'll take one more question. Then I think we'll get on with the program because we have a number of papers that are talking on the general subject of the problems in amplifiers. I see a hand back there?

MR. FORESCA, Cosbed Cable Division: To answer the gentlemen's question about the color of the sync bar. At one time I did some experiments on it. If you do change the cathode bias of the telebed that's involved in the intermodulation process, you will find that as you go with the bias on one side of the linear portion you will get one polarity sync pulse. As you go on the other side, you get a different polarity sync pulse.

MR. SCHLAFLY: Yes, I think that is correct.

MR. FORESCA: My question is that. To what extent the number of channels would affect the intermodulation level?

MR. SCHLAFLY: This is partially subjective, because it depends on where it lands. But if it synchronous it is an additive. I believe that it is an additive situation. Do any of you gentlemen care to comment on that?

MR. TAYLOR: This gentleman here has a question. Let's take one more.

UNIDENTIFIED SPEAKER: Mine is not a question. I just thought that maybe he might inform us as to how he set the CW signal on channel and then superimposed the other on the test?

MR. SCHLAFLY: As to how that was done?

UNIDENTIFIED SPEAKER: Just to tell us how you set this up?

MR. SCHLAFLY: There were 12 input channels and there was a CW generator. The input channel levels were set through those modulators and the CW level was set at the same level as the carriers that we had from off-the-air signal.

UNIDENTIFIED SPEAKER: But, when you set the CW signal in this test you showed, do you kill the other information on that channel?

MR. SCHLAFLY: Yes, sir. There was only the pure CW signal on that channel and it beat against the various signals of the television channels that were superimposed, mixed at the input to the amplifier. (Applause)

MR. TAYLOR: Again I want to thank Mr. Schlafly very, very much for coordinating this study on behalf of the Ad Hoc Standards Committee. And, again, I want to express the tremendous cooperation that has been given to this project by a number of manufacturers. We'll delay just a moment or two while the gear is removed to make possible some other visual aids.

I might say that many of the papers you will hear today are available in printed form and will be available on this table against the wall.

Our next speaker is Dr. Jacob Shekel of the Spencer-Kennedy Laboratory. You have his sketch here showing his background and I will not take further time to introduce him. Dr. Jacob Shekel. (Applause)

DR. JACOB SHEKEL: My talk will concern noise and cross-modulation from a few points of view. We all realize that noise and cross-modulation are the factors that ultimately limit the length of the system or the number of amplifiers that can be cascaded. But, I'm not sure that everybody here knows exactly how to estimate how many of the amplifiers can be cascaded and when that limit is reached; and how to do this before the system is built and before you find it out in effect. I want to separate the problem into three parts: First, how do we specify or measure or estimate the noise and cross-modulation of a single amplifier? Then, knowing that, how do we estimate the accumulation of noise and cross-modulation along the trunk line and distribution amplifiers? And, the third question, where do we stop? How far do we let it accumulate before we say this is as far as we go, because we cannot degrade the picture any further?

I am not going to discuss the third question. I am not going to give any numerical values on what should be the final noise or the final cross-modulation, because that is really up to the Standards Committee to set up. I don't think there is yet any complete agreement between the manufacturers or between the system users on the ultimate degradation that can be allowed. But I will describe what I hope is a very simple way of how to figure out from the specification of a single amplifier what the noise and cross-modulation of the total system are expected to be at the furthest point.

A simple way to estimate the noise at the end of the system, and one that I know is used quite extensively by system operators, is a simple measurement with a field-strength-meter. You measure the level of a certain channel at the furthest point of the line. Then you turn off the channel at the head-end and see what measurement you can read on the field-strength-meter. This measurement is due only to noise accumulated all along the system. You take the ratio of these two measurements--that is, the difference of the db readings--and this is what is called carrier-to-noise ratio in decibels.

Now, since this is such a simple method to measure the carrier-to-noise ratio of the system, we can also define and specify tne amplifier the same way. Suppose we take a single amplifier of the kind that we're using in the trunk line, and connect the field-strength-meter at its output, terminate the input and read the meter. Let's take as an example that the field-strength-meter reads at a certain channel --28 dbmv. Then suppose that this amplifier is specified to be used at an output level of +33 dbmv. By subtracting the two numbers, remembering that one of them is negative, the carrier-to-noise ratio of a single amplifier appears to be the difference between 33 and minus 28, which is 61 decibels. This, I think, is the simplest way to measure and to estimate the carrier-to-noise ratio of a single amplifier, a measurement that every operator can do right in his own office or in the field.

Knowing the carrier-to-noise ratio of a single amplifier, what can I expect to be the carrier-to-noise ratio when I cascade any number of them? Or, an alternative question, how many can I cascade if I want the carrier-to-noise ratio to be at least (let's say) 45 db?

Now, here we have to go a little into a table of decibels. I want to show a very simple method that every one of us can follow to make up his own table of decibels without reference to any handbook or any slide rules. I think it's a very handy thing to know.

First, we have to realize that the noise is a random waveform, and if you take the noise contributions of the various amplifiers they are not coherent. If you project them on a scope there will be no similarities between the noise waveforms of the various amplifiers. When such waveforms are added, the power of the total wave is equal to the sum of the powers of the various contributions. That means that a noise of two amplifiers will be 3 db higher than the noise of a single amplifier, and the noise in a trunk of 10 amplifiers will be 10 db higher than that of a single amplifier.

These are the only two numbers that we have to remember, that "twice" is 3 db and "10 times" is 10 db. I am going to write down the column of dbs from 0 to 10. We know that 0 db is a ratio of 1, and every time we add 3 db we double the ratio. Twice is the same as 3 db. So 3 db would be 2, and 6 db is 4, and 9 db is 8, and 12 db is 16, 15 db is 32 and 18 db is 64. Now, we go the other way. 10db is 10 times. Going backwards, 3 db below that, 7 db would be 5 times, and 3 db below that, 4 db, would be 2.5, and 3 db below that would be 1.25. To complete the table we now go sideways, multiplying and dividing by 10.

DB	NUMBER	DB	NUMBER
0	1	10	10
1	1.25	11	12.5
2	1.6	12	16
3	2	13	20
4	2.5	14	25
5	3.2	15	32
6	4	16	40
7	5	17	50
8	6.4	18	64
9	8	19	80
10	10	20	100

For each 3 db step: Multiply by 2 ↓ / Divide ↑

Multiply by 10 →

← Divide by 10

Fig. 1

So, now we know how noise of various amplifiers will combine, or how the carrier-to-noise ratio will change along the line . In our example I have used the carrier-to-noise ratio of a single amplifier at 61 db. If I had two amplifiers they will be 3 db worse, or 58 db; and with 10 amplifiers, it will be 51 db.

Now, let's take the following question:. If I start with a 61 db carrier-to-noise ratio of a single amplifier, how many can I cascade before I reach 45 db? The difference between 61 and 45 is 16. Going to the table, 16 means 40 amplifiers. So, 40 amplifiers of that type, operated at that level, will give a carrier-to-noise ration of 45 db. I can't say whether that is acceptable or not, but at least the system operator can go to the end of the system and measure the carrier-to-noise ratio, and if the reading is far from 45 db, then he knows right away that something must be wrong.

This is as far as we can estimate the carrier-to-noise ratio of a single amplifier and of a line with cascaded amplifiers. And, you will have noticed that I am trying here to specify the noise of the amplifier not by its noise figure, which is a certain measurement referred to the input, but by its noise output. First of all it is easier to estimate the output C/N ratio, and also it's a figure which is much easier to measure right in the field.

The second limitation on the system performance is the matter of cross-modulation. Now, of course, I'm not sure if we all know, after the previous demonstration, what exactly cross-modulation is. Maybe we know much more than we knew an hour ago. But, for the purpose of my talk it suffices that we can put a number to it. We say that an amplifier operated at a certain level with a given number of channels will have a certain amount of cross-modulation.

First of all, it is important that both the level and the number of carriers be specified, because the amount of cross-modulation changes with those two numbers, as it was demonstrated before. Also, the number which specified the cross-modulation can be given in two ways: It can be given in negative decibels (or "db down"), or it can be given as a percentage modulation. The meaning of the latter is that if we start with a CW carrier as our test signal, the modulation imposed by the other carriers will be a certain percentage.

The two specifications are equivalent to each other and there is a very simple way of passing from one to the other.

Let's look first at just the middle line of the nomogram on page 165, the one that is marked "cross-modulation". Here you see two scales, one in decibels and the other in percentage. For example, minus 40 db corresponds to 1%, minus 60 to .10% and minus 72 corresponds to .025%.

Now I would like to suggest that specifications be given in percentage rather than db, because then the way that cross-modulation accumulates along the trunk is very simply computed: You just multiply this number by the number of amplifiers. Let's take as an example that a trunk amplifier is specified to have .008% cross-modulation when operated at an output level of +33 dbmv with 12 channels. (How to get this number in the first place will be shown later. It could be the number given directly by the manufacturer, or it may have been computed from an equivalent number given by the manufacturer.)

The cross-modulation is really a superposition of the modulation of other channels onto the channel we are watching. And as we go along the line, all the contributions of all the amplifiers just add up in phase on top of each other, because all the channels progress along the line at the same speed. If we have a cross-modulation of .008% for one amplifier, we will have a cross-mod of .016% for 2 amplifiers and .024% for 3 amplifiers. Suppose we have 30 trunk amplifiers, and all of them operated at the same level, the total cross-modulation will be .008 times 30, which is .24%.

Now, this is only the trunk. We also add cross-modulation in a bridging amplifier, distribution amplifiers, line extension amplifiers. (Incidentally, in these amplifiers, since we try to operate them at the highest level possible, we do have cross-modulation, but we have almost no effect on the noise. That's why I have disregarded it in the first part of my talk).

We have .24% accumulated along the trunk line. Suppose that now we start from here into a distribution amplifier, and let's take again as an example that it is specified to have .1% cross-modulation at +58 dbmv for 12-channel operation. If we operate it at this level, it will add .1% cross-mod. Again all the channels come to this amplifier at the same time, all together, so on top of the .24% from the trunk line we have to add .1% of the distribution amplifier and we end up with .34%.

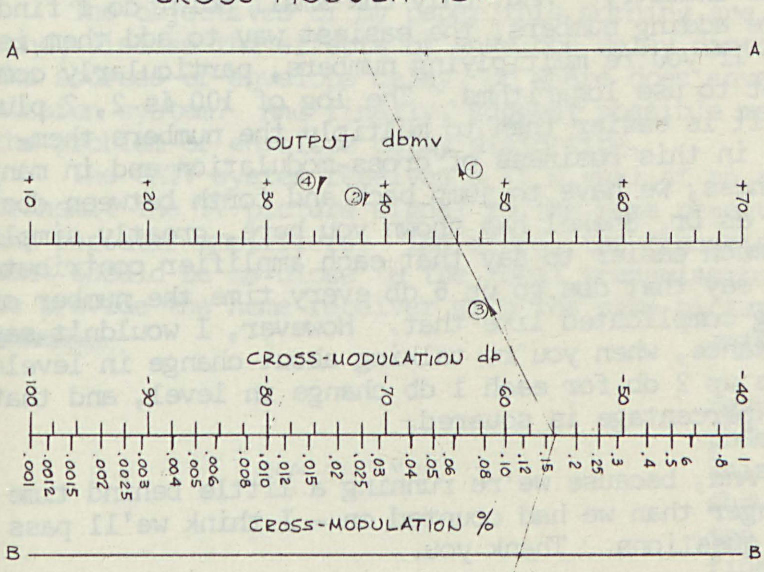
If we have no further amplifiers in the line, we can expect the customer to have a cross-mod of .34%. Suppose that a customer is further along the line, where we have another line-extension amplifier. All we have to do again is to add the cross-modulation contributed by that amplifier. And, thus by a simple process of addition of the contributions of various amplifiers we can very easily estimate the cross-modulation of the pictures at the customer tap.

There is some difference between noise and cross-mod measurements, because the noise can easily be measured at the customer tap and you can compare this to the computed results; whereas, the cross-mod measurement is a little more complicated and the equipment is usually not such that can be taken to the customers' house or carried around in a truck. I hope that within a year or so maybe some of the manufacturers will come up with small kits to measure cross-modulation, when the Standard Committee will have decided on a method that is satisfactory to everybody.

Now, the only thing that remains in this method of estimating noise and cross-mod is how to find the cross-modulation of a single amplifier. In noise, it was simple. We just take an amplifier and measure it. We disregard the manufacturer's specs; we can check it every time.

On cross-mod we have to start from one number given by the manufacturer; and various manufacturers have various ways of specifying. For example, some manufacturers specify the level at which the cross-modulation for a number of channels is 57 db, while at least one other manufacturer specifies the level at which the cross-modulation is .05% when only two carriers are used in the test.

NOMOGRAM FOR THIRD-ORDER CROSS-MODULATION



The nomogram shows a way of comparing these various specifications; and also a method to estimate what the cross-mod would be at the level that you actually use in the amplifier.

There are three scales on this nomogram: The middle scale is the cross-modulation that we discussed before. The upper scale is the output of the amplifier in dbmv, and the lower scale is the number of carriers used. This scale is really in two parts and we can use either of them, whichever is more convenient.

For a first example I'll start with an amplifier specified at .05% two carriers at +46 dbmv, and we want to know what would be the cross-modulation if you operated 12 carriers at +33 dbmv. First, we use the bottom part of the nomogram to find what is the effect of number of carriers. (Here I want to point out that the assumption on this nomogram is that the cross-modulation from various carriers will add incoherently. If we have many carriers that are on the same network that produce coherent sync pulses, the addition will be more severe than it is here.)

So, we join the point of "2 carriers," and .05% cross-modulation, find the intersection with line B, and then connect the "12-carrier" point through that intersection point up to the cross-modulation scale. This shows something around .16% (we don't really have to estimate this point exactly because this is only a partial answer).

Now, we go to the upper part of the nomogram and see what effect the level will have. We have now changed the number of carriers from 2 to 12. We take that intermediate point (which is roughly .16%), correct it to the +46 dbmv and go up to line A. From that point we come down to the +3? dbmv point and we end up on the cross-mod scale at .008%.

To summarize, I've used the bottom part to see the effect of number of carriers at the same output level; and the upper part for the effect of the output level at a constant number of carriers.

As a second example, suppose the amplifier is specified as having -57 db cross-mod at the level of +48 dbmv for 12 carriers. Here we don't have to change the number of carriers, and all we have to work with is the upper part of the nomogram. We connect the point at -57 db to the +48 dbmv, go up to line A. Now, suppose we are using the amplifier at a level of +37 dbmv, so we connect that point from scale A through the +37 and come up to .010% cross-modulation. This is the starting point for the computation of the trunk line.

Well, this is really all I wanted to show. How we estimate, or how we read the specification, or how we measure noise and cross-modulation with a single amplifier at the level we are going to work it, how the noise and cross-mod accumulate along the line and what we can expect as the final noise and cross-mod at the end of the line.

Now, are there any questions?

MR. KEN SIMONS: This really isn't a question. I'm cheating. I'm going to say two words. First, I want to thank Dr. Shekel for a very clear presentation of some facts that are long overdue in this industry. And, only one small point do I find that I would try to add. If you're adding numbers, the easiest way to add them is to add them, 100 plus 100 is 200. If you're multiplying numbers, particularly complex numbers, it's often convenient to use logarithms. The log of 100 is 2, 2 plus 2 is 4, 10 to the 4th is 10,000. It is easier than to multiply the numbers themselves. In the same way I believe in this business of cross-modulation and in many other facets of our community business, we have to jump back and forth between dbs and percent, and I believe we can, as Dr. Shekel has shown you here, greatly simplify the relationships involved. It's much easier to say that each amplifier contributed .1% cross-modulation than it is to say that dbs go up 6 db every time the number of amplifiers is doubled, or something complicated like that. However, I wouldn't say this worked all the time. For instance, when you're talking about change in levels, the amount of cross-modulation goes up 2 db for each 1 db change in level, and that's easier to say than to say that the percentage is squared.

MR. TAYLOR: Thank you, Ken. And, because we're running a little behind time - our demonstration took a little longer than we had counted on - I think we'll pass onto another paper without further questions. Thank you.

DR. SHEKEL: May I just give a not-answer to the not-question. I just want to defend in a couple of words the method of percentage, although as I say they are both equivalent and you can easily pass from one to another. First of all, it seems to me that percentage is easier when you have to add the contributions of various types of amplifiers. The cross-modulation of one is multiplied by the number of amplifiers in the trunk; and when you add the distribution amplifier, it is much easier using percentages.

As far as seeing the effect of output level on cross-modulation when it is in percent, one way would be using the nomogram, and the second way would be using the same table of dbs that I just invented 10 minutes ago. Because, you will check that if you increase the level in dbs, then you multiply the cross-mod by the number that is in the second column. But, again, as I said, there are many right ways of doing this thing and none is better than the other. Some are only more convenient. Thank You. (Applause)

MR. TAYLOR: Thank you, again, Dr. Shekel, very much. Maybe somebody will volunteer to be chairman of the Standards Committee. You can see the problems that arise in those deliberations.

Our next presentation will be on a subject that is somewhat new in this industry, Envelope Delay in CATV. Gaylord Rogeness from AMECO in Phoenix, Arizona, is our speaker, and his background, biographical sketch has been placed in your hands. Mr. Rogeness.

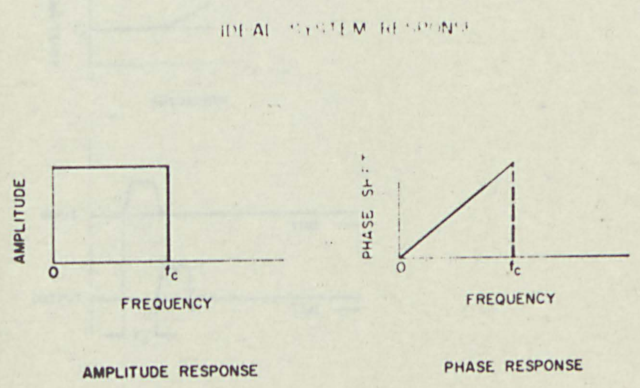
MR. GAYLORD ROGENESS: Thank you, Mr. Taylor. This morning I'm going to speak to you on the subject of Envelope Delay in CATV Systems.

Comparison of pictures produced by off-the-air signals and signals that have been transmitted through long cascades indicate that the off-the-air signal produces a sharper, more crisp picture. This effect is also more noticeable on low band channels compared to the high band channels. The low band channels produce a picture that is somewhat more fuzzy.

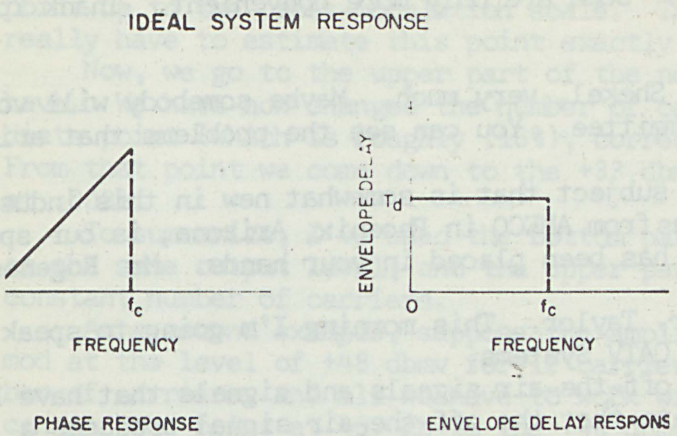
These effects exist even though the amplifier cascade has been aligned for optimum amplitude response, the cross modulation is at a minimum level, and the signal-to-noise ratio is high. Envelope delay distortion is a quantity which can explain some of these effects. Until recently, CATV systems have been providing pictures in areas where TV reception has not existed or has been very poor. Hence, there was little need to consider the more subtle transmission system requirements. However, as CATV moves into areas where competition with off-the-air reception exists, and the transmission of good color pictures is required, the effects of envelope delay have to be considered.

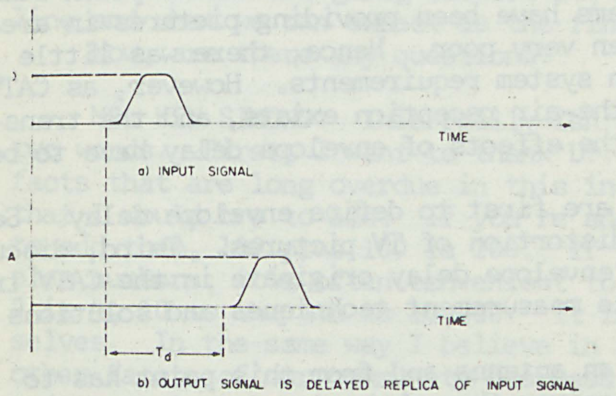
The objectives of my paper this morning are first to define envelope delay. Second, discuss the effects of envelope delay distortion of TV pictures. Third, discuss the sources of envelope delay, or where does envelope delay originate in the CATV transmission system? And finally, suggest possible measurement techniques and solutions to the problem of envelope delay distortion.

The CATV system receives a TV signal at an antenna and from this point has to transmit the TV picture signal to the home receiver through head-end equipment, cable and repeater amplifiers. Therefore, the transmission characteristics of this equipment should be as close to the ideal transmission characteristic as possible in order to provide the home receiver with the same picture quality that is received at the CATV antenna.



An ideal system has a flat amplitude response with respect to frequency and a phase shift characteristic that is linear. This is shown in FIGURE ONE. FIGURE TWO shows phase and delay characteristics of the ideal system. Envelope delay is defined as the rate of change of phase shift with respect to frequency. Or, in other words, envelope delay is the incremental slope of the phase shift curve versus frequency. In an ideal system the phase response is linear, so that the incremental slope of the phase response is constant. Hence, each frequency has the same value of envelope delay. It should also be noted that in an ideal system, time delay and envelope delay are equal.

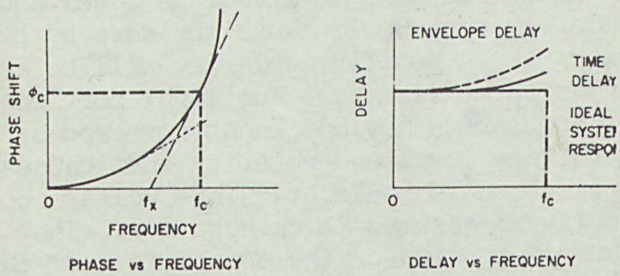




A TV picture signal consists of a sum of pulses which in turn are the sum of many frequency components. When this signal is transmitted through an ideal transmission system, each frequency component experiences the same delay. As a result, the TV picture signal at the output of the transmission system is the same as that at the input but delayed in time. FIGURE 3A shows a pulse applied to the input of a CATV system. If the ideal characteristic of flat amplitude and linear phase over the band of frequencies being transmitted exists, the output will be a delayed replica of the input as shown in FIGURE 3B. The output pulse waveform will be exactly the same as the input pulse waveform and will occur at a later point in time.

The difference between envelope delay and time delay is shown in FIGURE FOUR. These quantities are compared at the frequency f_c. Time delay is the phase shift at this frequency divided by the frequency, whereas envelope delay is the slope of the phase response at the frequency f_c. Note that the magnitude of envelope delay is larger than the time delay magnitude.

RELATION BETWEEN TIME DELAY AND ENVELOPE DELAY



$$\text{TIME DELAY} = \frac{\phi}{f_c}$$

$$\text{ENVELOPE DELAY} = \frac{\phi}{f_c - f_x}$$

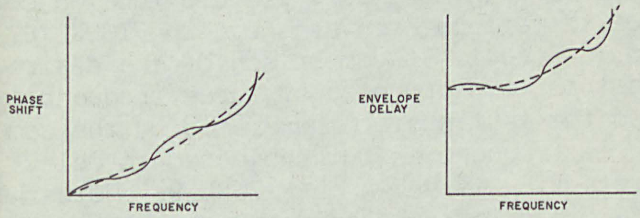
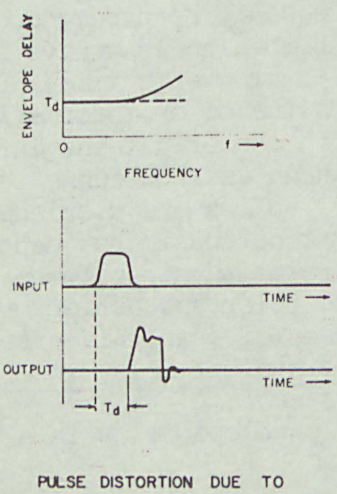


FIGURE 5. PHASE AND DELAY DEVIATION FROM LINEARITY. (NOTE THAT THE DOTTED LINE INDICATES THE SLOW DEVIATION FROM LINEARITY)



PULSE DISTORTION DUE TO NONUNIFORM ENVELOPE DELAY

FIGURE 4B shows a rough comparison of envelope delay and time delay over the frequency range of interest.

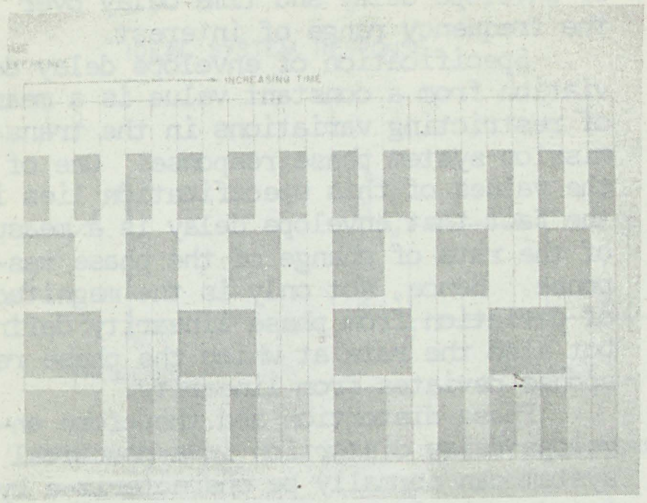
Specification of envelope delay deviation from a constant value is a means of restricting variations in the transmission system phase response. One of t the values of this specification lies in the fact that envelope delay is a measure of the rate of change of the phase response. Hence, not only is the magnitude of deviation from phase linearity defined, but also the rate at which the phase response deviates from linearity.

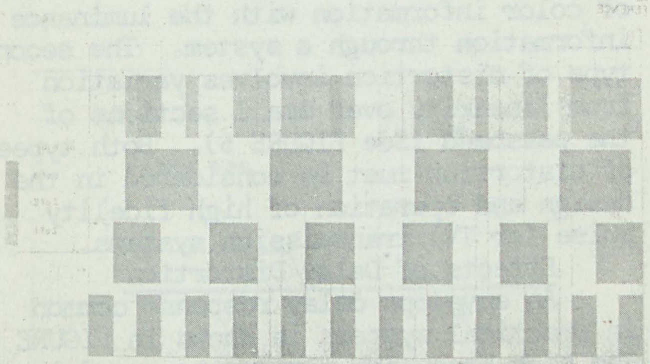
Phase distortion and therefore envelope delay distortion in a practical system can normally be characteriz=e by two descriptions. One is a gradual deviation from the linear phase characteristic, occurring over the major portion of the system passband. This distortion is important when comparing the transmission of color information with the luminance information through a system. The second type of distortion involves variation from linearity over small sections of the passband (See FIGURE 5). Both types of distortion must be considered in the design and operation of high fidelity pulse (or TV) transmission systems.

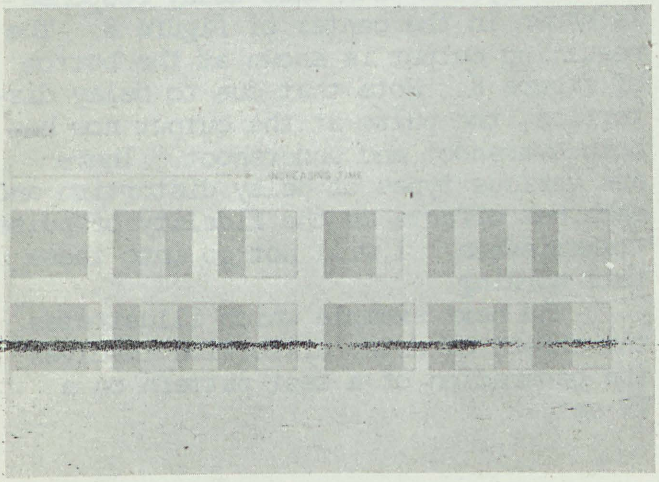
Effects of Delay Distortion

An envelope delay response common in practical systems is shown in FIGURE 6. The high frequency components are delayed by a greater amount than the low frequency components. A pulse applied to the transmission system with this delay response is shown in the center of figure 6. The resulting output is shown at the bottom of figure 6. Note that due to delay distortion, the pulse at the output now has both overshoot and undershoot. There are various types of delay distortion and each has effects on the fidelity of pulse transmission. I will not go into these this morning.

The next example which illustrates the effect of delay distortion involves the generation of a test pattern on a TV set.







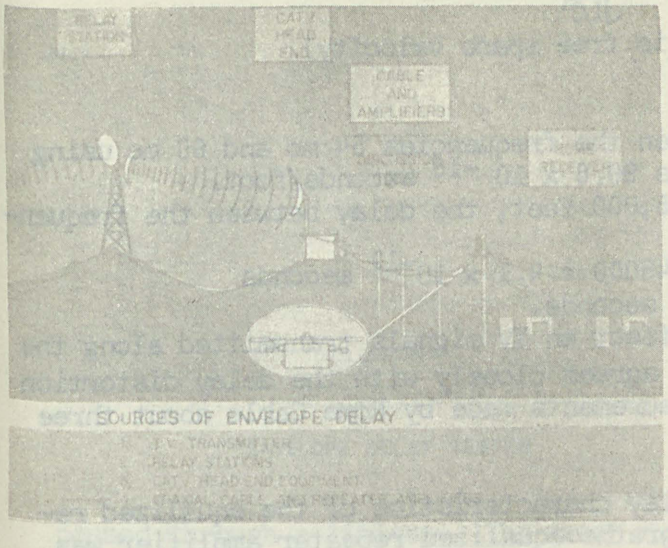
The bottom line of FIGURE 7 shows the desired test pattern. This test pattern is generated by the sum of three frequencies occurring in the time phase shown. The black portion of each frequency component shown corresponds to a voltage level and polarity that would cause the screen to be dark. A positive polarity will be assumed for this case. The light portion of each frequency component corresponds to a voltage level and polarity that causes the screen to be light. This voltage polarity is assumed to be negative. The darkest portion of the composite test pattern is then generated when all three positive voltages add at the same time. The dark gray is produced when only two positive voltages add. A completely white bar is produced when the negative voltages of all three frequency components add at the same time.

When all three frequency components are not delayed by the same amount during transmission to the TV picture tube, a composite test pattern as shown in FIGURE 8 could result. A comparison of the desired test pattern produced by three frequencies and the test pattern generated by the same three frequencies but subjected to delay distortion is shown in FIGURE 9. Note that the distorted pattern does resemble the desired pattern.

Consider next the effects of time delay distortion on a color picture. The color picture is composed of two main signals - the chrominance information which contains color information and the luminance signal which contains the brightness information. These signals are transmitted in different parts of the frequency spectrum, so it is important that both signals arrive at the TV picture tube at the same time. Due to delay distortion the color information may not coincide with the brightness information and an effect known as the "funny paper effect" occurs. Colors are displaced to the right or left of the image, depending upon the delay relationship between the picture carrier and color subcarrier. The red color is most sensitive to this effect.

Sources of Envelope Delay in a TV Transmission System

Sources of envelope delay in a TV trans-



mission system are depicted in FIGURE 10. A responsibility of the TV station is to transmit TV program material over the air. In so doing, the TV signal passes through equipment which have amplitude and phase characteristics that are frequency dependent.

Next a relay station may be necessary before the CATV system received the signal. This relay station is a second source of distortion in the system.

The CATV system consists of head end equipment, coaxial cable, and equalized amplifiers. Each of these three items is a potential source of distortion.

The signal finally arrives at the home receiver where it is processed and displayed. Many sources of amplitude and delay distortion exist in a TV set that is not properly aligned.

The FCC regulates the characteristics of the color TV signal being transmitted. The TV transmitter must have a prescribed envelope delay characteristic. This delay characteristic is specified to compensate for the delay distortion produced in the frequency selective circuits of the home receiver. The manufacturers of TV receivers use the specified delay characteristics of the transmitter to set design and manufacturing tolerances on their TV sets. Therefore, any picture transmission equipment placed between the TV transmitter and home receiver must be near perfect in order to minimize distortion.

Phase characteristics of the coaxial cable and equalized repeater amplifiers used in CATV systems will be discussed at some length today.

Phase Characteristic of Coaxial Cable

The transmission of energy along a coaxial cable is defined by the complex propagation constant. The propagation constant has a real and imaginary component. The real part describes attenuation along the cable and the imaginary component defines the phase shift constant of the coaxial cable. The propagation constant is

$$\gamma = \sqrt{(R+jwL)(G+jwC)} \quad (1)$$

For low loss cable, such as that used in the CATV industry, it is possible to simplify equation one and write the phase shift constant

$$\beta = \sqrt{LC}\left[1 + \frac{1}{8}\left(\frac{R}{WL}\right)^2\right] \quad \text{radians/unit length} \quad (2)$$

R, L., and C are the cable resistance, inductance, and capacitance per unit length and w is 2π times the frequency in cycles per second.

Remembering that envelope delay is the rate of change of phase shift with respect to frequency, the derivative of equation 2 yields the cable envelope delay.

$$T_E = \frac{dB}{dw} = \sqrt{LC}\left[1 - \frac{1}{8}\left(\frac{R}{WL}\right)^2\right] \quad \text{seconds/unit length} \quad (3)$$

Note that envelope delay is not constant with frequency because of the $\left(\frac{R}{WL}\right)^2$ term. However, the magnitude of this deviation from a constant value is small enough to have negligible effect. A numerical example will show this:

Constants taken from a cable manufacturer's data sheet for 75 ohm Alucel 1/2" coaxial cable are

Capacity $C = 16.5$ pf/foot

Velocity of Propagation $V_c = 0.82\ V_o = \frac{1}{\sqrt{LC}} = 7.87 \times 10^8$ ft/sec

Attenuation (V_o is free space velocity)

$\alpha = 0.006$ db/ft at 54 mc

$\alpha = 0.0065$ db/ft at 60 mc

The difference in envelope delay between the frequencies 54 mc and 60 mc using the cable constants listed and equation 3 is 90.6×10^{-18} seconds/foot.

For a 30 amplifier cascade extending 45,000 feet, the delay between the frequencer 54 mc and 60 mc is

$$T = 90.6 \times 10^{-18} \times 45000 = 4.1 \times 10^{-12} \text{ seconds}$$

$$T = 4.1 \text{ micro-micro seconds.}$$

This delay distortion has negligible effect on TV signals transmitted along the cable. This number of 4.1×10^{-12} seconds agrees closely with the delay distortion calculated from velocity of propagation measurements made by Rome Cable about three years ago.

<u>Repeater Amplifier Delay Characteristics</u>

The next problem is to describe the delay characteristics of the equalized repeater amplifier. A theoretical response for the equalized repeater amplifier was postulated for an 18db length of cable. The amplifier response was assumed maximally flat at both the low and high end. The high end roll off was assumed more steep than the low end because of the cut off characteristics of the transistors.

The transfer function of an equalized amplifier can be written:

$$\frac{e_{out}}{e_{in}} = \left[\frac{1 + j\frac{w}{w_1}}{1 + j\frac{w}{w_2}}\right]\left[\frac{\left(j\frac{w}{w_3}\right)^n}{1 + \left(j\frac{w}{w_3}\right)^n}\right]\left[\frac{1}{\left(1 + j\frac{w}{w_4}\right)^m}\right] \quad (4)$$

To calculate the envelope delay of this expression the phase response is first derived and is

$$\text{Phase} = \tan^{-1}\left(\frac{w}{w_1}\right) - \tan^{-1}\left(\frac{w}{w_2}\right) + 90° = \tan^{-1}\left(\frac{w}{w_3}\right)^n - \tan^{-1}\left(\frac{w}{w_4}\right)^m \quad (5)$$

The envelope delay is the derivation of equation 5 with respect to w ($2\pi f$).

$$T = \frac{1}{f_1}\left[\frac{1}{1+\left(\frac{w}{w_1}\right)^2}\right] - \frac{1}{f_2}\left[\frac{1}{1+\left(\frac{w}{w_2}\right)^2}\right] - \frac{n}{f_3}\left[\frac{\left(\frac{w}{w_3}\right)^{n-1}}{1+\left(\frac{w}{w_3}\right)^{2n}}\right] - \frac{m}{f_4}\left[\frac{\left(\frac{w}{w_4}\right)^{m-1}}{1+\left(\frac{w}{w_4}\right)^{2m}}\right] \quad (6)$$

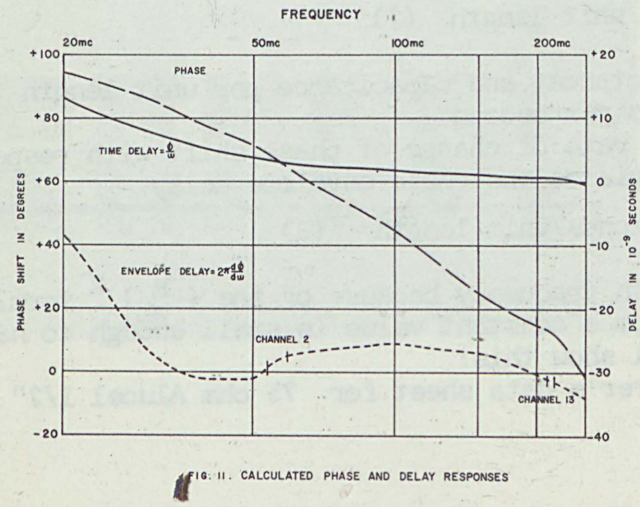


FIG. 11. CALCULATED PHASE AND DELAY RESPONSES

Phase, time delay, and envelope delay were calculated as a function of frequency using equation 5 and 6 are shown in FIGURE 11. The following values were used in the calculations:

$w = 2\pi f$

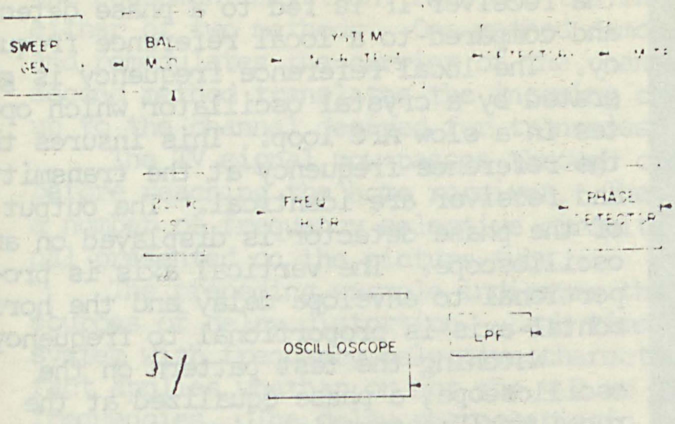
$w_1 = 2\pi \times 49.5 \times 10^6$ $n = 2$

$w_2 = 2\pi \times 334 \times 10^6$ $m = 4$

$w_3 = 2\pi \times 40 \times 10^6$

$w_4 = 2\pi \times 250 \times 10^6$

Note that envelope delay is not constant with frequency as is required for an ideal transmission system. Also note that channel 2 is more susceptible to response irregularities than channel 13 because it occupies a higher percentage bandwidth.



ENVELOPE DELAY TESTER

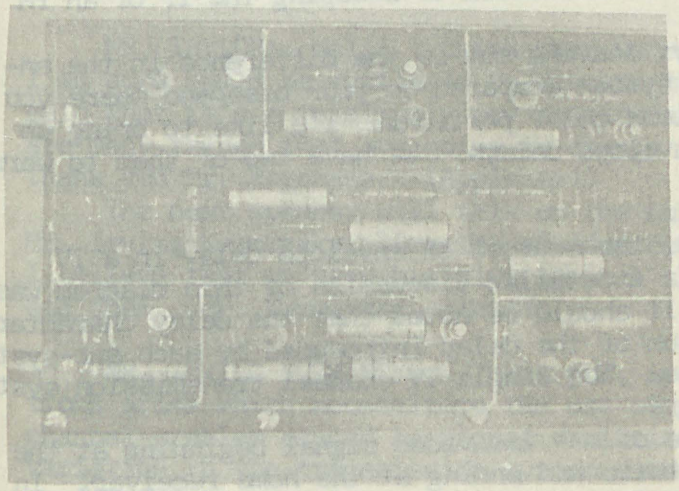
(6 mc bandwidth at 54 mc compared to 210 mc).

Envelope Delay Testing

A block diagram of a test set that measures envelope delay is shown in FIGURE 12. The 200kc reference oscillator output is applied to a frequency doubler and balanced modulator. The second input to the balanced modulator is a sweep generator. The output of the balanced modulator is two frequencies spaced at twice the reference oscillator frequency. These two signals are applied to the system under test and are swept across the frequency spectrum maintaining a constant spacing.

The test signals are detected at the output of the system under test and then passed through a limiter. The test signal at the output of the limiter is then compared in a phase detector with the output of the frequency doubler. Each of these signals is at the same frequency. However, the doubler output has a constant phase reference while the signal passed through the system under test is measuring the incremental slope of the system phase response. The output of the phase detector is a DC voltage proportional to the envelope delay of the system under test.

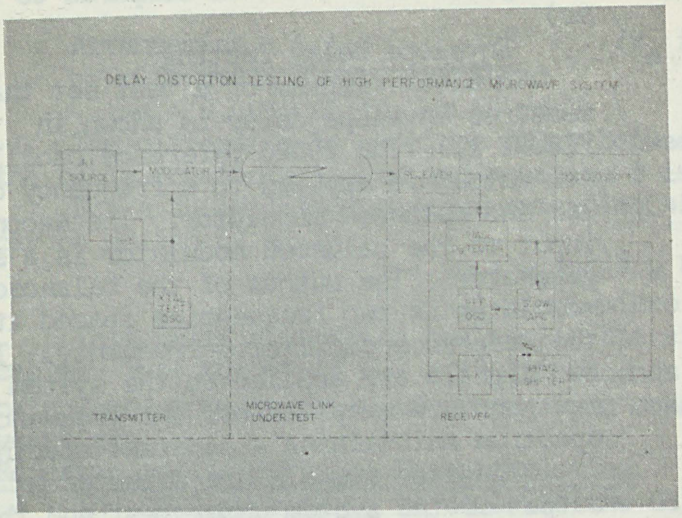
The oscilloscope displays envelope delay on the vertical axis and frequency on the horizontal axis. The vertical scale can be calibrated in terms of electrical degrees or directly in units of time (microseconds or nanoseconds). A frequency marker can be inserted into the test set for calibration of the horizontal scale.



An envelope delay test set was constructed by utilizing the principles described in the preceding three paragraphs. This test set is shown in FIGURE 13. Unfortunately, time did not permit the completion of many delay measurements before the convention. However, the envelope delay of a cascade of three AMECO ATM-70 amplifiers and 75 db (220mc) of coaxial cable was measured. The envelope delay characteristic was constant from 40 mc to about 90 mc and then began gradually sloping through the high band. The difference in delay across any high band channel was less than three nanoseconds (3×10^{-9} seconds).

The purpose of this next example is to point out that phase distortion, and hence delay distortion, can be measured and corrected in the field even though the transmitted and received signals are physically separated by large distances. The block diagram of a test set used by a manufacturer of microwave equipment to measure delay distortion of a microwave link is shown in FIGURE 14 (next page).

The crystal reference oscillator operates at about 500kc and modulates the RF source. The reference oscillator frequency is divided down to provide a sweep voltage to sweep the RF source through the passband of the transmission system. The swept RF signal is transmitted over the microwave link and is received at the remote



location of the receiver. The output of the receiver IF is fed to a phase detector and compared to a local reference frequency. The local reference frequency is generated by a crystal oscillator which operates in a slow AFC loop. This insures that the reference frequency at the transmitter and receiver are identical. The output of the phase detector is displayed on an oscilloscope. The vertical axis is proportional to envelope delay and the horizontal axis is proportional to frequency.

Watching the test pattern on the oscilloscope, a phase equalized at the receiver IF output is adjusted until the transmission system delay distortion is minimized.

Today I have defined envelope delay as the rate of change of the transmission system phase response. Some of the effects of delay distortion on the transmission of TV pictures were mentioned as a loss of crispness of the black and white signal and a funny paper effect on color pictures.

I believe that we must now develop test equipment to accurately measure the CATV system delay characteristics. After the delay characteristics have been measured, phase and/or delay equalizers can be designed to compensate for existing delay distortion. Thank you. (Applause)

MR. TAYLOR: Thank you very much. I think we would have time for one or two questions if somebody would like to. I see one here.

MR. WILLIAM CRUZ, (Collins Radio): I think it should be important at this point and time with your fine speech here to separate the distinction of envelope delay of your sweeping, the RF spectrum where cable activities -- it's all very proper, very correct. I agree with you completely. Your discussion of sweeping the IF of an FM or microwave system is also correct.

One other thing you are, sorry to say, leaving out is the difference in the envelope delay of your RF system or your IF system compared to your baseband where you are considering the envelope delay of various color portions. I'd like to bring up the point that they are quite different envelope delays. We have two of them to worry about. Thank you.

MR. ROGENESS: Referring to figure 10, the sources of envelope delay in a TV transmission system are shown pictorially. A detailed discussion of this diagram was not made because of the time limitation. It should be noted that the delay characteristic of a linear system is equal to the sum of the delay introduced by each sub-system contributing to the overall system response. The single TV channel transmission system delay characteristics are of importance here.

As an example, follow the transmission of a 4.2mc video signal beginning at the output of the TV camera at the broadcast studio and ending at the home receiver. In the transmission of this 4.2mc video signal from the broadcast studio to the home receiver, the video signal will be translated a number of times. For example, the video signal at the broadcast studio is mixed or translated to an RF frequency for broadcast. A microwave relay station may then receive this signal, translate it to IF frequencies and amplify it, and then mix back up to a microwave frequency for transmission at microwave. The next relay station may then translate the signal from microwave back to RF frequencies for re-transmission.

At this point a CATV system may receive the signal off the air. The CATV head-end equipment may then translate the incoming channel to a different channel by either of two methods. One method demodulates the incoming TV channel to baseband and remodulates the carrier of the channel to be transmitted over the cable. A second method translates the incoming channel down to IF frequencies and then back up to the channel desired for transmission over the cable.

The TV signal now passes through coaxial cable and equalized repeater amplifiers before reaching the home receiver. The TV signal in the home receiver passes through a number of frequency selective circuits before it is demodulated and the video signal presented on the picture tube.

The preceding example indicates that the 4.2 mc video signal is subjected to many sources of delay distortion. Each time it passes through a network or transmission system with frequency selective characteristics, delay distortion is possible. This fact applies whether or not the 4.2 mc TV signal is at video, IF, RF or microwave frequencies. The delay characteristic of the TV transmission path between TV camera and the TV picture tube in the home receiver is equal to the sum of the delay characteristics of each frequency selective network that the TV signal passes through.

The CATV system has control of the transmission characteristics of the head-end equipment and the cable system. Therefore, from a knowledge of the delay characteristics of the transmission systems external to the CATV system and a knowledge of the overall delay characteristic required to transmit an undistorted TV picture, the CATV system delay response can be specified.

MR. SABIN FLORESCU, from Carlsbad Cable Division: We were talking about envelope problems in the RF transmission systems, just the same way Bill Cruz said it. Our biggest problems are in the modulators. What do we do about them?

MR. ROGENESS: There are two types of phase distortion. One is differential phase which is a cross modulation of the color and luminance signals and is a function of the nonlinearity of the modulator; whereas envelope delay -- or the characteristics I was talking about were related to the phase response of the transmission system which are constant.

The delay response of head-end equipment between CATV antenna and coaxial cable must be constant with frequency in order to solve Mr. Florescu's problems.

MR. TAYLOR: Well, I think that we're running a little behind time. Mr. Rogeness I am sure would be available to discuss this question. I think it can also be safely said that it's a relatively new consideration in our industry and I am sure there are many things that are going to change in the future as a result of this discussion. Thank you Mr. Rogeness. (Applause)

The next speaker will talk on the subject of "Automatic Gain Control in CATV". Mr. Irving Kuzminsky, Director of Advanced Product Engineering of Entron, Inc. And, I believe that we have his biographical sketch to circulate if the pages will circulate them. Mr. Kuzminsky, please.

MR. IRVING KUZMINSKY: Thank you, Archer. In a CATV system, two types of situations arise which necessitate the use of gain control. One is a narrow-band single-channel problem caused by signal variations at the antenna. The other is a wide-band variation in the transmission system caused by changes in either the cable or the amplifiers.

In order for the system to function properly, it is necessary to first eliminate the variations in signal level which are normally encountered at receiving sites. Let us consider what might happen at the customer's receiver if this were not done.

Most present day CATV systems utilized adjacent-channel transmission as a means of most efficiently carrying the maximum number of channels at a minimum cost. However, as far as the receivers are concerned, the adjacent channels are potential sources of interference. This was the reason that, in the early days of CATV, some people thought that adjacent-channel systems would not work. In order for these systems to work properly, it is necessary to accurately control the levels of the signals with respect to each other so that the receiver is able to pick out the selected signal without objectionable interference from other signals.

Once the single-channel signals are combined onto a common line, random variations of these signals would be impossible to handle. This is because the gain of the trunk amplifiers is controlled on a wide-band basis. That is, the gain is varied in a coherent manner to all channels in the amplifier passband simultaneously. With random variation of each channel's signal, cross modulation and noise problems would be encountered in the trunkline system. With some signals going up, some going down, and others remaining constant, gain control would be impossible, and the problems generated are obvious. Thus, stabilization of the antenna signals is mandatory before the signals are inserted into the trunk system.

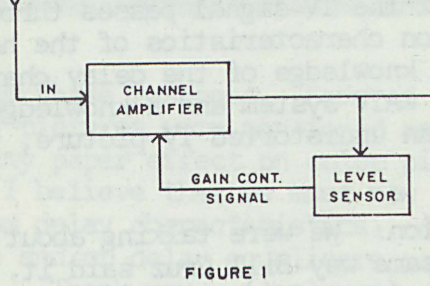


FIGURE 1

The variation in antenna signal level is usually handled by the method shown in FIGURE 1. The signal is amplified in a single-channel RF amplifier. The output signal is detected and provides a DC control signal which is indicative of the output signal level. This control signal is used to vary the operating point of the intermediate stages and, by this means, the gain of the amplifier so as to maintain the output at a nearly constant predetermined level.

FIGURE 2 is a plot of the output level of a typical single-channel AGC amplifier. The amplifier being considered has a gain of 60 db. Curve A indicates that, with no AGC there is a linear relation between input and output except for high levels where the amplifier overloads. Curves B, C, and D show that, for small signals, the output follows the input. However, once the AGC threshold is exceeded, the output remains almost constant. Thus, for proper AGC operation, a minimum signal level is required depending on the setting of an output level control.

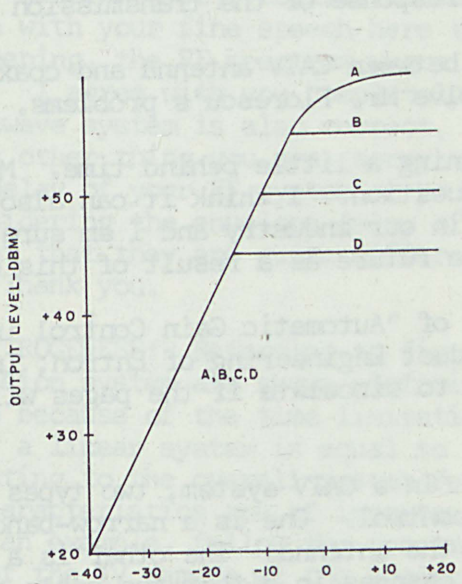


FIGURE 2

This is called "delayed AGC" because gain control is delayed until the threshold signal is reached. Curves B, C, and D represent different delays. The maximum allowable input level is determined by the overload characteristics of the amplifier.

Normally, the input and output stages are not varied, since varying the input stage affects noise figure and input match, and varying the output stage affects the overload level of this stage. Because of these noise and overload limitations, some other method should be used where large signal level variations exist.

Consider the block diagram shown in FIGURE 3. The RF input signal is amplified and detected. When the detected signal exceeds a predetermined amplitude a delay is activated and an attenuator is inserted between the antenna and the head end equipment. When the signal decreases sufficiently, the attenuator is removed. A cascade of four such switchable attenuator sections--each section having 10 db attenuation-- effectively reduces a 60 db signal swing to 20 db. This smaller swing can then be handled by the AGC arrangement previously considered.

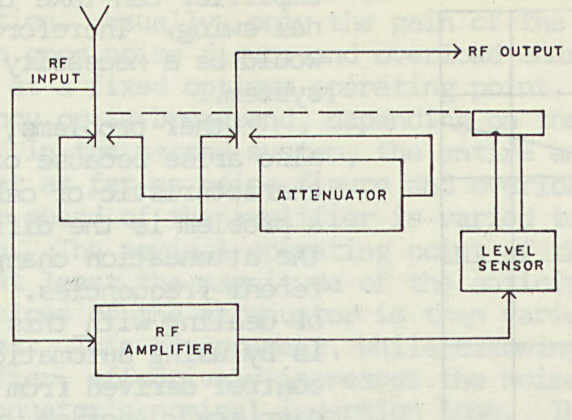


FIGURE 3

FIGURE 4 is a typical plot of output level vs. input level for a four-section controller. "A" is a plot of output level vs. input level with no compensation and, and, of course, the changes in output level follow the changes in input level. The output level vs. input level is shown by "B" for increasing, and by "C" for decreasing signal. At any given level, the input can vary over a 20 db range with no switching occurring. For example, at Point 1, with an input of 18 dbmv, two attenuators have been switched in so that the output is 18 - 20 or -2 dbmv. As long as the input signal level remains between +10 and 30 dbmv, no switching will occur, and operation will be along the joinging Points 2 and 3.

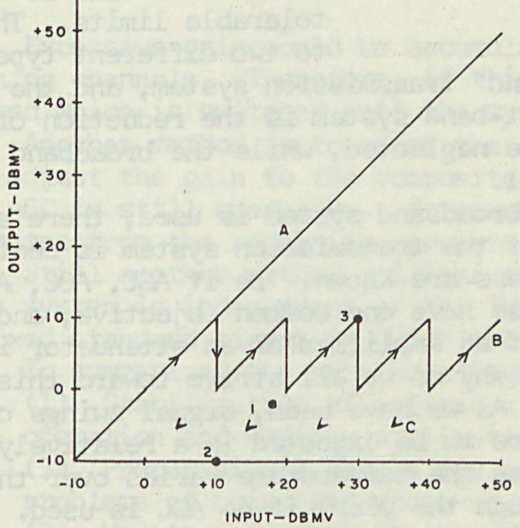


FIGURE 4

Once the signal levels at the head end are stabilized, the signals are ready to be inserted into the transmission system. Since the signals are stabilized, why is AGC necessary in the trunkline amplifiers? To answer this question, it is necessary to look at the entire trunkline system. While the signals may be stabilized at the input to the trunkline, they will still vary in the trunkline because of changes in cable attenuation with temperature variation and because of changes in amplifier gain. While the latter factor is a matter of conjecture, the change in cable attenuation is a well known fact and can be predicted.

If the last amplifier at the end of the longest trunk is capable of handling the largest signal swing expected then AGC is not required. FIGURE 5 (next page) shows the correction factor which must be applied to the 68° value of cable attenuation to obtain the attenuation at some other temperature.

We can see that the extreme temperature to which the cable may be subjected, attenuation correction factors are obtained of 1.06 at +120°F and 0.90 at -20°F. This means that for each 100 db of cable attenuation, there results an increase of 6 db at 120°F and a decrease of 10 db at -20°F.

A trunkline consisting of 1/2 inch foam dielectric aluminum jacketed cable may typically have an attenuation of 1.3 db per 100 feet at Channel 13 at 68°F. In a five mile line, this would amount to 340 db attenuation. However, at 120°F this would increase by 20.4 db, and at -20°F it would decrease by 34 db. No presently existing amplifier can take this signal swing. Therefore, AGC would be a necessity in this system.

Other problems, of course, also arise because of this characteristic of cable. Such a problem is the difference in the attenuation change at different frequencies. One way of dealing with this problem is by using automatic tilt control derived from two pilot carriers. Another way of dealing with the problem is to use thermal equalizers. A third way is to split the band so that the accumulated difference in cable attenuation within each band is within tolerable limits. This leads to two different types of transmission systems, the so-called "Split-Band" transmission system, and the broadband transmission system. The advantage of a split-band system is the reduction of change of tilt effects to the point where they may be neglected, while the broadband system must use some method of automatic tilt control.

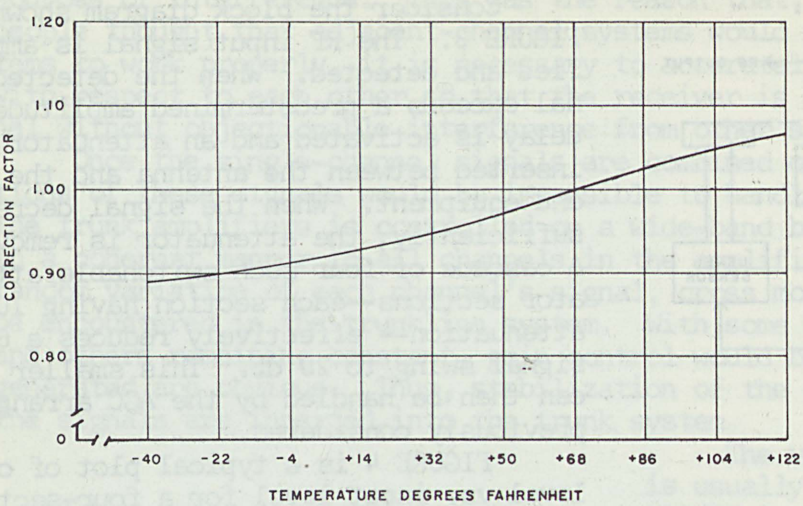


FIGURE 5

FIGURE 5

Regardless of whether a split-band or a broadband system is used, there are still many methods in use today by which the gain of the transmission system is controlled. There are also many names by which these systems are known. Be it AGC, AOC, ALC, AVC, or A--you name it--C, all of the methods in use have one common objective, and that is to vary the gain or loss (in some cases) of an amplifier or an attenuator in an attempt to maintain a constant signal level. Why do we all strive toward this goal?

As we have seen, signal swings of 20 or 30 db are to be expected in a relatively short system as the temperature varies over the day and through the year, if no AGC is used. The use of AGC reduces maintenance problems by eliminating the need for periodic resetting of levels. Too, compensation for cable and equipment aging is provided to some extent. Let us look at some of the different methods that are used to achieve these goals.

Whether the transmission system is broadband or split-band, operation of the AGC circuits in either of these systems may be controlled by either TV signals or by pilot carrier signals. Also, either a single signal or a multiplicity of signals may be used for AGC purposes. Thus, many possible types of AGC systems exist. However, all of these methods are very similar in actual operation. FIGURE 6 is a block diagram which illustrates two methods which might be used with either TV signal or pilot carrier AGC systems.

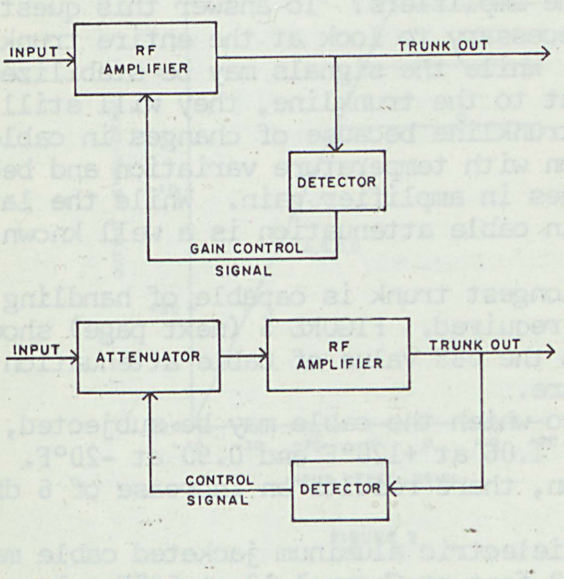


FIGURE 6

FIGURE 6

In the first case, the amplifier is operated below its maximum gain capability so as to be able to compensate for anticipated changes in input signal in either direction. Usually, only the gain of the intermediate stages are varied so as to maintain good noise figure and overload characteristics while the input and output stages are at a fixed optimum operating point. The detector is tuned either to a single frequency or is broadband, depending on the type of AGC system used.

In the second system, the entire amplifier is maintained at its optimum operating point as far as noise figure and overload characteristics are concerned. The attenuator ahead of the amplifier is varied to change the overall gain at the amplifier station. The nominal operating point of the attenuator must provide an insertion loss of at least the magnitude of the anticipated downward change in input signal level. The loss of the attenuator is then varied up or down to correct for changes in input level. This arrangement, while allowing optimum operation of each stage of the amplifier, effectively increases the noise figure of the amplifier by the amount of the attenuator's nominal insertion loss. The best solution may be a combination of the two methods. That is, place the attenuator at an intermediate point in the amplifier. This would allow optimum operation of the active elements in the amplifier and, at the same time, provide a good noise figure.

As stated previously, the AGC may be derived from either TV signals or from pilot carrier signals. A single TV signal cannot be used alone to activate the AGC because if that channel went off for any reason, all amplifiers would run wide open.

Excessive gain would be accumulated, and overload would soon occur on the remaining channels. Therefore, if this method is used, a standby oscillator is required which is switched into the system if the primary source goes off.

Another method is to sense the composite signals in the passband of the amplifier and adjust the gain to the composite level. With this method, if a station goes off, the AGC is still operative. No standby oscillator is required since the AGC circuit operates from the remaining carriers.

Still another mtthod utilizes only pilot carriers to drive the AGC circuits. This system is independent of the TV signal levels and has the advantage of providing a fixed standard signal to which the entire system may be referenced.

To summarize, the main advantages of AGC are:

(1) Stabilization of individual channel signals permits adjacent channel operation and maximum utilization of the transmission system.

(2) Proper signal levels may be maintained in the trunk, thereby avoiding problems of noise and cross modulation.

(3) Maintenance problems are reduced by eliminating the necessity to reset levels with changes in temperature.

Thank you. (Applause)

MR. TAYLOR: I think we can take time for one or two questions. Anybody have a question they want to ask Mr. Kuzminsky? One in the back of the room.

UNIDENTIFIED SPEAKER: This might be going back to this envelope delay problem, but I notice on the color set there was another image to the right and I've had this problem on black and white. I don't know what it is. Is it miss match?

MR. KUZMINSKY: Well, sounds like it.

UNIDENTIFIED SPEAKER: Miss match?

MR. KUZMINSKY: Yes.

MR. TAYLOR: Thank you. Thank you very much, Mr. Kuzminsky. (Applause)

Our next speaker is Mr. Robert Cowart, Vice President in Charge of Construction for Viking Company. And, he's going to talk on "System Reliability". I believe the sketches will be circulated while he's talking. Mr. Robert Cowart.

MR. ROBERT COWART: I like in particular that ETA part, so I'll be very brief. I'd like to talk to you this morning briefly on system reliability.

These days we find outselves building more and more systems into areas that already have available to them strong high quality, highly reliable, local off-the-air signals. In order for a system to compete under these conditions, the system must be engineered in such a fashion that it could successfully compete in terms of the same quality, reliability and performance as these off-the-air signals. The preliminary determinants of quality are signal-to-noise ratio, cross modulation and ghosts. You are all familiar with these terms as a result of the industry schools which outline and detail methods of qualitative determination. I am sure that by now you are all familiar with these terms, with their method of determination and know of many ways in which to improve them. A fourth, extremely important, factor is reliability. A subject which has frequently been ignored both in the past and at the present. My purpose today is to acquaint you with the basics of reliability and to point out to you some methods by which present system reliability can be improved.

Many studies have been made in the past both by military and commercial interests in the pursuit of those factors that control and influence reliability. In almost every case explored the most highly reliable system was the simplest system. I am sure you will all agree from your own experience that this is the case. The military answer for increased reliability is redundancy. This means having almost two complete sets of basic equipment, one ready to take over the function of the first, should it fail. The commercial solution to reliability is primarily by increasing the reliability of the components and increasing the size, weight and mass of the device. This is more or less the brute force approach.

In CATV, neither of these two standard approaches is really available to us because of the unusual demands we make of the device. In transistorized equipment we sacrifice virtually everything for the sake of a lower noise figure or increased output capability. We are pushing the upper limits of the State of the Art. We can't use redundancy because of cost. We can't use higher reliable components because high reliable, high performance transistors are not available yet. We must achieve our reliability in the method in which we construct our systems, and in the method in which we utilize the manufacture's product.

Most manufacturers today design equipment that is inherently reliable. In many, many cases that we have examined, we find that this inherent reliability of the device is lost in its application.

Reliability in electronics systems is generally considered to mean the length of time between events that render the system incapable of performing its designed function. In industry, exhaustive and extremely expensive studies are made to determine and assign quantitative values for the time between failure. This period is often referred to as mean time between failure or MTBF. In CATV these numbers are not available but the principle guiding the establishment of these numbers is available and it is with this principle that we will concern this discussion.

If all of the components of an electronics system are considered to be functionally in series and if the failure of any components in this series chain results in a system failure then the overall system reliability can be expressed by a very simple formula. This formula states that the overall system reliability, designated by the symbol "R", is equal to the reliability of each of the series components raised to the power of the number of those components that are in series.

$$R = r^n$$

Where r = mean reliability (probability function) of each component.
n = number of components in series.

This expression demonstrates something that you know intuitively to be true. In other words, the longer your trunk line in a system the greater the probability of failure of a component of the trunk. Conversely, the shorter the line the less chance of failure. The formula also allows us to show mathematically that given two different amplifiers if twice as many amplifiers are used in a system of Type "A" as are Type "B" and Type "B" has half the reliability of Type "A" then the overall reliability of the system is exactly the same because there are twice as many pieces used but the reliability of each piece is twice as great. You intuitively know that the statement is correct.

The formula also shows that the high reliability system would have few parts and each part in itself should have the highest possible reliability. Towards accomplishing this end we customarily, in large systems, use extremely low loss trunk cable such as 3/4" aluminum and the highest possible db spacing between amplifiers because in our trunk system the highest reliability component is the cable; secondly, would undoubtedly be the connector; thirdly, the accessory items, splitters, directional couplers, etc.; and lastly with least reliability is the amplifier itself. Our major significant contribution to reliability of that trunk segment would be to decrease, by whatever means we could the cable loss, utilize wide amplifier spacing, etc., the number of amplifiers functionally in series. In our efforts to increase the reliability of that trunk segment we would attempt to reduce the total number of objects with lesser reliability than the cable to a minimum. This would mean we would reduce the number of splices, if possible, by care in our construction; we would reduce the number of splitters, directional couplers, equalizers and other objects inserted in the lines and try and make as much of the line as we could, sheer cable: Because, of course, the cable is the most highly reliable item of our components.

The same reasoning establishes a guide line in the design of equipment and has prompted most major manufacturers to abandon the practice of using splitters to generate inputs to associated distribution equipment and to instead build into the trunk amplifier chassis a fixed directional coupler to provide the input to distribution. When this is done we eliminate a jumper and several connectors that we used to use in the past to accomplish this. The same reasoning demands that in transistorized equipment the equipment should be mounted without equipment enclosures. That means not with the use of an equipment cabinet. When an equipment cabinet is used the signal must pass through a bulkhead connector, a mating connector internally in the cabinet, a jumper, and finally through another connector on the end of the jumper and into the amplifier chassis. The same thing is true on the output of the amplifier. When this is done there are five additional elements functionally in series with the signal between the two ends of the trunk cable. Although connectors have inherently high reliability, by removing the eight connector assemblies from the line and replacing them with two direct entry connectors, we have thus improved the reliability of each amplifier station four times. You intuitively know that the reliability of this configuration is far less than the direct entry type connector permanently mounted to the amplifier chassis.

In an operating system when you examine at the end of the year the maintenance that has been given to the system, you find some rather curious things. You find first of all that many of your system outages were not caused by any inherent failure of the amplifier itself. You find that they were caused by such unrelated things as power failures; by cars breaking off power poles; by trees falling across distribution and trunk cables; by the failure of fuses as a function of temperature; by lightning strikes; and by employee carelessness in leaving amplifiers disconnected, etc. Another important point that gains in significance as we move into the area of transistorized system construction with many, many, amplifiers dependant on a single power supply is that extreme caution should be used in selecting the location for the power supply. I am sure that you have all had an experience where a certain amplifier in your system continually caused you trouble because of failure of secondary voltage delivered by the power company. We have seen amplifiers installed and taking power from power company transformers already seriously over-loaded. Few of you have given any thought to requesting the power company to provide you with your own transformer, which need not be very large, to assure yourself of a non-interrupted source of power.

The cost is very low and the reward in terms of increased reliability is great. These things again point up the fact that in system design, a system should be engineered in such a fashion so that the absolute minimum of active elements of the system are in cascade. Ideally, as we have all discussed many times in the past, a system would be arranged in the manner of a wheel; with the center of the wheel the point of signal origination and of radial lines from the wheel hub to the outlying distribution areas. Although this is obviously impractical in most cases, an attempt to accomplish this type of construction can be made by the adoption and usage of extremely low loss master trunk cables as a backbone of the system. This new configuration will resemble somewhat the skeleton of a fish; with the master trunk cable being the backbone of the skeleton and distribution at right angles to this master trunk but in much, much smaller segments.

Many of you have suggested in the past that you accomplished a form of redundancy by paralleling master trunks perhaps several blocks or half a mile or so apart, but when you examine the situation existing in parallel trunk, you find that you have not accomplished your purpose because the basic law of system reliability catches up with you. Remember, it states that the reliability decreases exponentially in proportion to the number of active elements in cascade. By paralleling master trunk you are in effect doubling the number of elements in cascade. Now it is true that by the redundant parallel trunk method you do restrict the service fault to a smaller area, however, if the two or more trunk segments are exactly the same length, then the system reliability itself, on the basis of our definition of fault, is actually impaired by the same number of trunk lines existing.

In summary, let's recap the major points that we have established. A system gains RELIABILITY by SIMPLICITY. This means that when you make your new layouts, look at them carefully to determine if you have taken the shortest route, if you have arranged your construction to utilize a minimum of connectors and splices, see if your power feeds come from a reliable source and make sure that you are utilizing as fully as possible the reliability delivered to you by the manufacturers.

Thank you. (Applause)

MR. TAYLOR: I would open the floor to questions on any of the subjects we've been answering. Let's take any questions to Bob Cowart first, if you have them, however. Are there any questions on this Systems Reliability that you'd like to ask Bob Cowart? Well, if there are no questions specifically to Bob, they may arise later. I shut off a number of questions earlier, particularly on the subject of envelope delay. Ken Simons.

MR. SIMONS: Again, this isn't a question. I would like to take a few minutes if you don't mind to show you a little scheme that we have used for some years in our lab to measure group delay. You might call it Do-it-yourself-group-delay-measurement. It takes equipment that most of you will have in your service shop or lab and I think there's enough time to sketch it out. The accuracy is not of the highest order but perhaps we'll make up for that in the cheapness of the equipment used.

I should give a credit here. The very fine grease pencil I'm about to use is through the courtesy of VIKING.

Now, the basis of this method of group delay measurement is the constant delay of a long piece of cable. If you have a reel of cable and I'll represent it this way. That's a piece of cable. It's on a reel and you're looking at it in 4th dimension. The delay from here to here is constant, approximately constant as Mr. Rogeness told us. How we can use this constant delay thing to help us in measuring the delay error or the actual group delay of a piece of equipment? Well, we start over here with a sweep frequency generator. There are a good many reputable manufacturers - you can take your choice.

We split two ways with either a 6 db resistive splitter or a 3 db reactive splitter. We have now two outputs, one going here and one here. This one goes up to the cable and now over here we combine again. And, we do nothing here. We just put in a jumper, very simple. Over here we put a detector and we put it on the scope.

Now, because we have two pads here combining at this point, we have reinforcement at frequencies where the two signals are in phase. We have cancellation at frequencies where they are out of phase and net result: We have a pattern which is this one and if the cable is quite long it does it quite often. High frequency ripple pattern you might call it.

Because the delay of the cable is constant, the ripples occur at precisely spaced frequencies and a very simple relation exists at the frequency separation between adjacent minima and gives us the group delay.

The frequencies in microseconds and then we have microseconds. If the frequency is in megacycles, the time delay is in microseconds. The only limitation the method has -- it has two limitations. You want a good piece of cable that has a very uniform and impedance characteristic just so that various anomalies don't get in. The loss of the cable -- no -- the accuracy of balance. I left one item out of here. You have to put an attenuator in because the signal at this point is reduced by the loss of the cable, -- a low loss cable works better and you put in a certain amount of attenuation just so that the minima gets sharp, the nose are sharp.

Now, having very sharp nose, you can very precisely determine this frequency and take their difference and get the group delay. You first do this with just the cable and get a delay characteristic of the cable, which is approximately constant, surprisingly constant in practice, and then you put in whatever you want to measure here - system or amplifier. It adds delay to this lag and ripples become more closely spaced where the group delay is steep and space outward isn't. So you have a group delay characteristic, calculated and plotted on paper. And I don't think it costs anything providing you have all the equipment. (Applause)

MR. TAYLOR: Thank you, Ken. I see a question right here. Will you identify yourself, please?

MR. DONALD LEVINSON, Wheeling, West Virginia: I would like to know whether anybody in the industry has been using the AT&T VIT signals and whether any work has been done in their field to evaluate our systems?

MR. TAYLOR: Does anybody care to respond on that? Will you identify yourself?

MR. BOB LEWIS, Dubuque, Ohio: The problem using these VIT Signals you end up taking tronoscope or scope for the time delay. I used it for checking microwave and I used it for taking times 24 scope. It is a good indication of color response. You can use it on the system. We've done this, but it takes a wide band detector, so there's problems in doing it but it is a good check.

MR. TAYLOR: Thank you. Anybody have further questions? Thank you all. Meeting is adjourned.

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