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Orthogonal Frequency Division Multiplexing: An Application to High Definition Television

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FIGURES

Orthogonal Frequency Division Multiplexing: An Application to High Definition Television

George Hufford*

"Orthogonal Frequency Division Multiplexing" is an interesting system architecture for high data-rate digital transmission that has been receiving renewed attention. This report is a discussion of how it works and of how it might be used to carry High Definition Television (HDTV). The principle tool is a computer simulation that allows us to look at many aspects of an OFDM system. It is concluded that the claims for the system largely hold up, and that its use might simplify many technical problems for HDTV.

Keywords: computer simulation; HDTV; interference; multipath; OFDM; signal-to-noise; spectrum efficiency

1. INTRODUCTION

Over-the-air radio transmission of high data rates must contend with noise, interference, multipath, and whatever other imperfections of the propagation channel are present. One interesting suggestion for a remedy involves a change in the signal architecture, using a technique sometimes called "multiple carrier modulation" but more often (and we shall follow custom here) *orthogonal frequency division multiplexing* (OFDM). In this architecture, a stream of symbols is broken up into blocks and each symbol of a block is used to modulate a separate carrier. If there are, say, N different evenly spaced carriers then the blocks will be of length N, and if we are to transmit a given information throughput then the modulation rate on the separate carriers may be reduced to $1/N$ times the original rate. Not only has there been no change in the overall rate of transmission but there is also no change in bandwidth requirements. If we use a bandwidth B and assume that this will support the "single carrier modulation" transmission of symbols at the rate B, then the individual slow-rate channels will transmit at the rate B / N and have a bandwidth equal to B / N—thus the total used bandwidth is again B.

The presumed advantage of such a system lies in the reduced rates of the individual channels and their consequent resistance to multipath problems. The individual bandwidths are now narrow enough that all fading can be characterized as "flat," and inter-symbol interference should be negligible. And perhaps of equal importance, there is a new improvement possible in how the spectrum needs are managed: the narrow bandwidths mean tight skirts that can crowd upon channel edges, and the many carriers mean some flexibility in where they are placed, thus perhaps helping to solve intersystem problems.

The scheme is not a new one—the original papers [1,2] date from 1957 and 1961. But there has been a renewed interest in its advantages, particularly in Europe where

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its application to digital audio broadcasting [3,4] has been implemented. In that case multipath conditions can be severe and will change rapidly, thus making a tactic like OFDM almost a necessity. The scheme is to accept signals from a number of broadcasters, and to interleave these signals to make a combined high data rate which is then broken into blocks and sent to the multiple carriers.

This ability to interleave information symbols is another feature of OFDM. It leads to a kind of frequency diversity that can be used to combat frequency selective fading— symbols have faded in one part of the spectrum, perhaps related symbols using other frequencies will allow recovery. And this kind of reasoning brings up a still further point. The appearance of blocks of digital data is a very natural place to introduce error-correcting block codes—say of the Reed-Solomon type. When this is done, we speak of COFDM: *Coded* orthogonal frequency division multiplexing. It can provide a virtually error-free transmission under quite adverse conditions.

The present report is a study of OFDM and how this sort of signal architecture might be used in the transmission of a single high data-rate broadcast signal such as digital television. In particular, present plans for High Definition Television (HDTV) call for a digital signal. It will need about 20 Mbit/s, and will be restricted to the present 6-MHz television channels, implying symbols that carry 4 or 5 bits of information.

Our principle tool in this study is a computer simulation in which we have modeled at least the first order behavior of the system and of possible effects from the environment. Much inspiration for this effort was obtained from work performed at the Canadian Communications Research Centre (see, especially, [5]) and to some extent, what we have tried to do here is to duplicate some of their results. Indeed, it was one of the original purposes of our project to prepare a straightforward computer program that would provide a separate confirmation of results obtained by the commercial simulators that are certainly more sophisticated but also less transparent in their operations. This purpose seemed well fitted to the present study of OFDM and its application to HDTV.

2. BACKGROUND THEORY

To describe the OFDM signal more precisely, let us suppose we have an incoming stream of symbols represented as complex numbers in some sort of constellation such as that from a quadrature amplitude modulation (QAM) scheme. We divide these up into blocks of length N and write them in the form

$$z_{j,n}, \qquad j = 0, \ldots, N - 1$$

Thus for each n we have a complex-valued vector of dimension N and our data set has become a sequence of such vectors. These are now used to modulate a group of carrier frequencies giving the combined complex signal

$$s(t) = e^{-i2\pi\nu_c t} \sum_n \sum_{j=0}^{N-1} z_{j,n}\, p\left(t/T - n\right) e^{-i2\pi j\beta t} \tag{1}$$

where ν_c is a "carrier" frequency, $p(x)$ is a rectangular pulse of length 1, T is the time interval used to transmit each vector, and β is the frequency separation between the succession of multiplexed carriers. This is just a linear combination using the symbol values for coefficients, with the elementary functions

$$u_{j,n} = p(t/T - n)e^{-i2\pi j\beta t} . \tag{2}$$

These are presumably orthogonal for different n because the pulses do not overlap in time and for different j because they do not overlap in spectrum. Thus the word "orthogonal" in the name OFDM, and thus the supposition that a receiver will be able to easily separate the symbols. Actually the functions do overlap in spectrum—for a given n, the inner product that determines orthogonality may be written

$$(u_{j,n}, u_{k,n}) = \int_{-T/2}^{T/2} e^{-i2\pi(j-k)\beta t} dt = T \operatorname{sinc}(\pi(j-k)\beta T) \tag{3}$$

and this vanishes for $j \neq k$ if and only if βT is an integer. One normally assumes this quantity equal to 1, but later we shall deliberately make it somewhat larger. The total bandwidth is approximately $B = N\beta$ and the information transfer is $F = N/T$ (symbols per second). We would expect $F \approx B$ so that once again $\beta T \approx 1$. In the case of HDTV, B is constrained to 6 MHz, so that most of the remaining parameters are almost fixed. If we set $N = 512$, then β is approximately 12 kHz and T approximately 83 μs.

One convenience offered by OFDM is in the way the collected signal can be formed for transmission. The idea follows from noting that (1) is just the Fourier series of something. Then suppose we sample the signal s(t) at regular intervals τ. Looking at only one of the time blocks and ignoring the carrier frequency υ_c, we have

$$s(k\tau) = \sum_{j=0}^{N-1} z_j e^{-i2\pi jk\beta\tau} \tag{4}$$

which looks just like an inverse DFT (discrete Fourier transform) provided, as we shall suppose, $\tau = 1 / B$ and $\beta\tau = 1 / N$. Then to form the OFDM signal, we assemble the $z_0,\ldots,z_{N-1}$, compute the DFT as

$$Z_k = \sum_{j=0}^{N-1} z_j e^{-i2\pi jk/N} , \tag{5}$$

arrange these into a sequence of pulses

$$\hat{s}(t) = \sum Z_k \delta(t - k\tau), \tag{6}$$

pass this through a filter with rectangular frequency response from $-B/2$ to $B/2$, and finally multiply by the rf carrier. The result will be exactly the function in (1).

Conversely, to receive the correct bit stream we need only sample the incoming signal and perform a direct DFT. This provides the z_j immediately. Note that things seem turned around. The z_j are the "spectral" components for the time series Z_k. Of course,

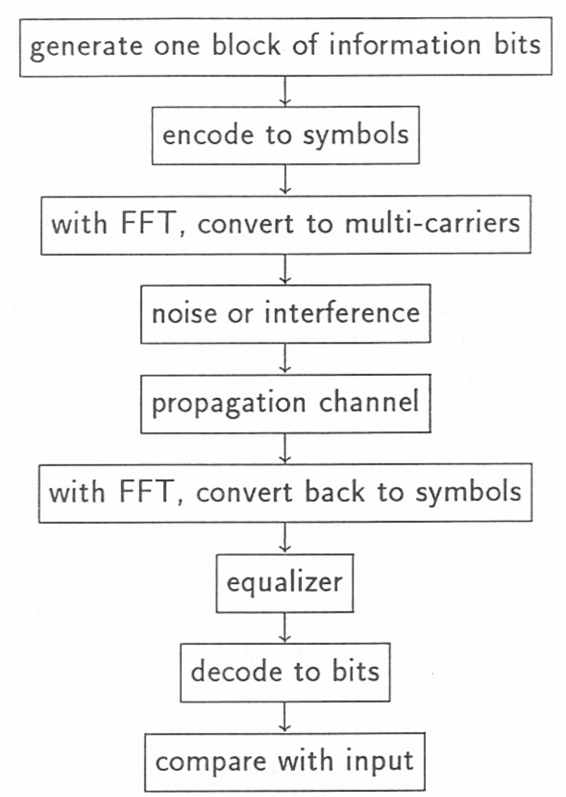

Figure 1. A flow diagram of the simulation program.

one will probably want to arrange numbers so that the DFT's can be implemented by a radix-2 FFT (*fast* Fourier transform), which can in turn be implemented by digital signal processors.

It is also a feature of OFDM that the transmitted z_j need not all be information symbols. Some might be synchronizing signals or calibrating signals. And some might be simply set to zero to avoid broadcasting on that frequency or having to receive on that frequency. Indeed, if the chosen N is not a power of 2 and one still wants to use a radix-2 FFT, one can use the next value that is a power of 2 and simply "pad out" the sequence with zeroes. On reception the extra carrier frequencies can just be ignored.

3. SIMULATION

Our simulation efforts were for a standard desktop computer using FORTRAN. Basically, the process flow is, as shown in Figure 1 (above), quite straightforward. It proceeds one block at a time and operates entirely in the time domain. Signals are modeled in the baseband as complex-valued functions of time—the physically real signal being, then, the real part of this baseband signal after it has been multiplied by an exponential at the carrier frequency. Most of the modules in Figure 1 have several implementations and which one is used is a choice that is to be made at run time. This is how we have allowed for a variety of system setups.

The generated information bits are pseudo-random and come from a maximal length shift-register sequence with 31 stages. Its initial state is controlled by the hardware clock

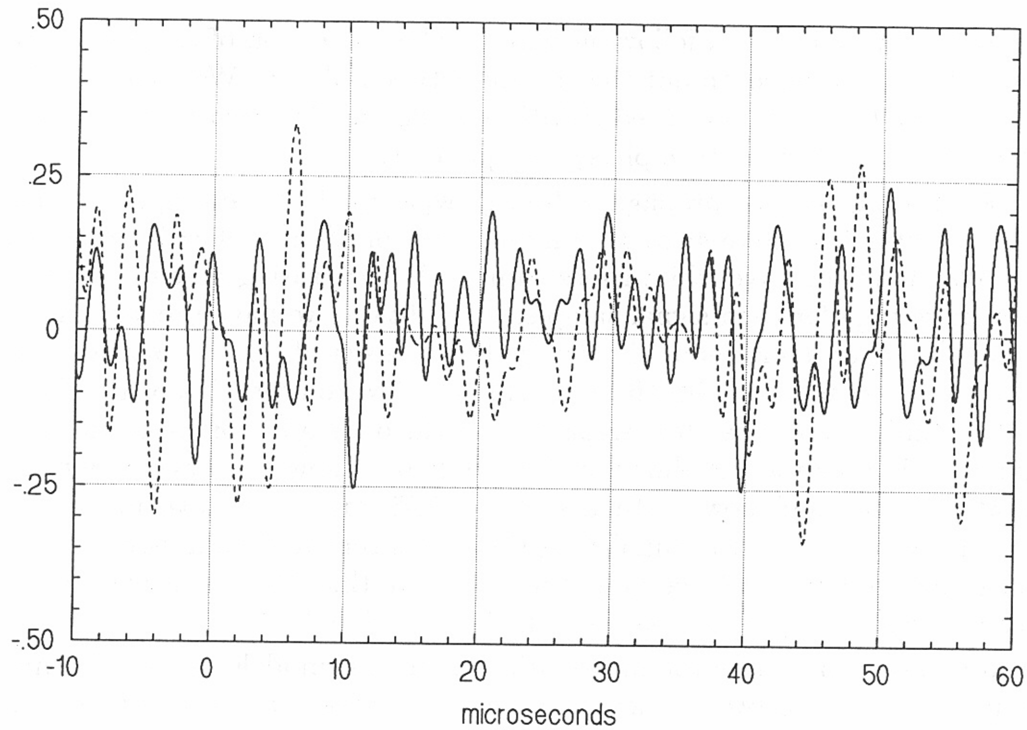

Figure 2. The complex-valued baseband signal of a simulated transmission of a little more than one block of symbols using OFDM with BPSK modulation. The imaginary (quadrature phase) component is the dotted curve. A complete block starts at 0 and ends at 50 μs.

and should be pretty random. But note that other random numbers—those of a real-valued nature—are generated by an entirely independent subprogram. For this we have chosen a "modular additive" approach suggested by Knuth and reported in [6, chap. 7].

The process of going from bit stream to symbols comprises the first step in the signal modulation. We have written several modules for this—from the simple BPSK (binary phase-shift keying) to the more involved 32TCM (trellis-coded modulation)—and one can imagine devising many others. It is at this point that one might also have introduced an error correcting block code, but for our present purposes we have left out this step.

Since 32TCM provides 4 fairly robust bits per symbol it seems like a good candidate for the HDTV signal. The particular version we have implemented is the simplest of the suggestions given by Ungerboeck [7]. It entails a 5-bit cross-shaped constellation (as displayed in the complex plane) in which one of the bits is reserved for parity. The "trellis" refers to the Viterbi algorithm used to provide a partial amount of error correction.

When the sequence of symbols is prepared, the conversion to a multi-carrier modulation is fairly straightforward. To make the results appear more like an analog signal, however, we assume a bandwidth perhaps 4 times what it needs to be, extending the input signal with zeroes. Then the subsequent time samples will be separated by a δt that is 1/4 the necessary value, and the signal will form a band-limited interpolation of what a real transmitter would have produced. An illustration of such a signal is given in Figure 2 (above). This shows the complex-valued baseband signal for one complete block along with portions of the preceding and following blocks. The succession of signals has passed through a low-pass filter to smooth out the change between blocks. We have set $N = 64$, $B =$ 1.28 MHz, $T = 50$ *fLS,* and have used BPSK modulation. The two traces are the real (co-phase) and imaginary (quadrature phase) components.

At this point the signal must propagate through what might be an imperfect channel. There is the possibility of additive noise or of interference or of multiplicative noise caused by multipath. We model additive white gaussian noise by generating a new, independent, complex gaussian voltage for each signal sample. In addition, we have prepared files that simulate different kinds of interference. Presently, they consist of, first, a periodic train of impulses and then an example of a line from an NTSC television signal. For this latter we have chosen the standard color bar test signal whose video form is portrayed in Figure 3. The spectrum as it is broadcast is shown in Figure 4; note how the lower sideband has been greatly abbreviated and how, with a spike at 5.75 MHz, we have introduced an (unmodulated) audio signal. In the simulation the line is repeated endlessly without the expected interuption by a vertical blanking interval. Note that in normal operation such interference signals will be asynchronous with the OFDM signal.

Multiplicative noise caused by the propagation channel is modeled as a finite impulse response. At present we have two of these routines. The first is a simple two-ray model consisting of the direct ray followed by a second which can be of any selected size and delay time and which will be given a random phase. The second follows the scheme in [8] in which the direct ray is followed by a tail of exponentially decreasing "uncorrelated scatterers." An example is shown in Figure 5. This is a fairly modest amount of multipath, and the sampling interval corresponds to what was used in Figure 2. Note that the time scale implies overly high frequencies that must be removed in the subsequent analysis.

Following through the flow of Figure 1, the signal now will have arrived at the receiver. It is passed through a low pass filter much as the band pass filter it would have passed through in a real receiver, although now the process also covers some of the smoothing that should have taken place in the transmitter. The filter is implemented by a simple finite impulse response (FIR) system. It is followed by a sampling process carried out at the proper OFDM rate, and then an FFT produces what we expect to be, barring transmission impairments, the original symbols. In Figure 6 is an illustration of the results at this point. It shows a 128-carrier sequence of quadrature phase-shift keying (QPSK) symbols after it has passed through a pure, unimpaired, channel. The rounded corners at beginning and end show the effects of the low pass filter.

After the equalizer (which we describe later) the symbols are decoded into a sequence of bits and the result is compared with the original input bits. At this point also, one can prepare a display showing various aspects of the processes. Many of the figures in this report were prepared in this way.

And so we have reached the end of the processing procedure for one OFDM block. If desired, one can now loop back to process a second block. To compute BER's (bit error ratios), for example, the usual approach is to let the process repeat over and over again until the count of errors reaches a pre-determined number. If this number is n, if the errors appear independently of the others, and if the BER is small, then the BER can be estimated to be n divided by the total number of bits. The estimation will have a relative

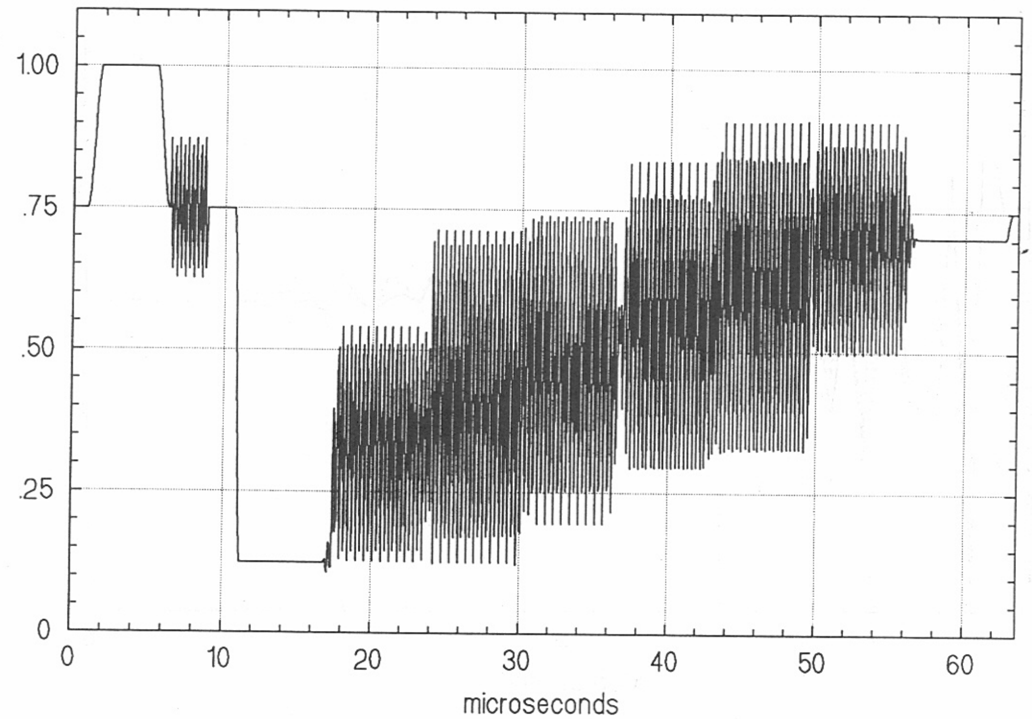

Figure 3. The video signal of a line from the color bar test signal for NTSC television.

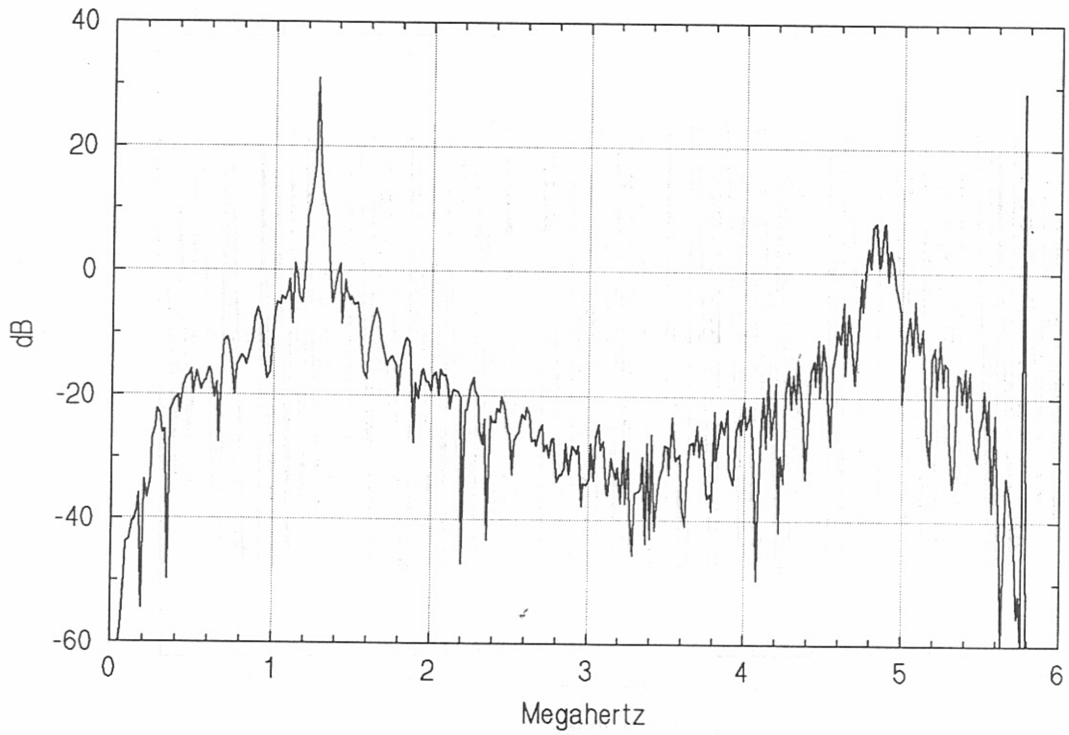

Figure 4. Spectrum of the broadcast signal of the color bar test line in Figure 3.

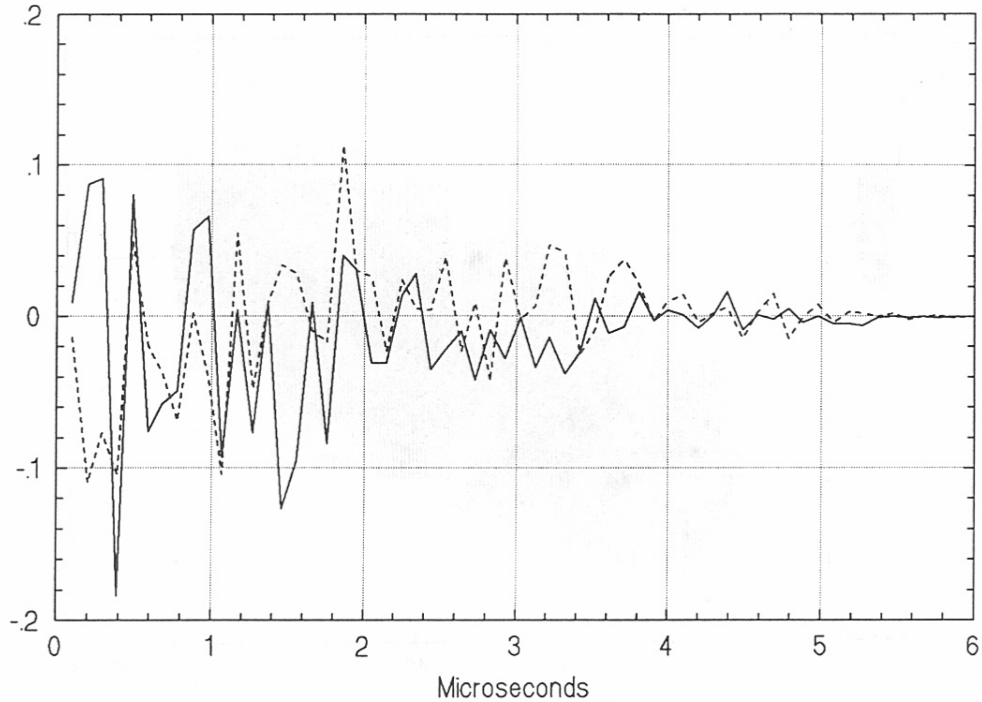

Figure 5. Illustrating the complex-valued tail in the impulse response using uncorrelated scatterers. The decay time is 1 μs, and the strength relative to the direct ray is -6 dB.

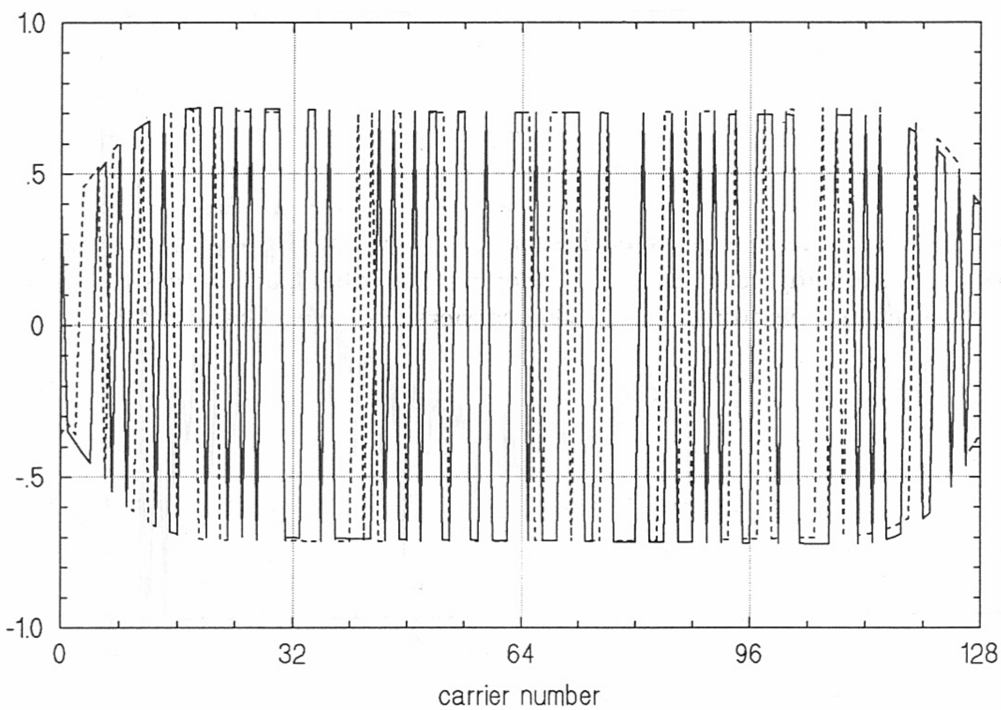

Figure 6. The received symbols from a single OFDM block using QPSK. There are 128 active carriers.

error whose root mean square is approximately 1 / $\sqrt{n}$. Thus, even if n is as small as 10, results should at least have the right order of magnitude.

In the next section, the BER computations were all done with $n = 40$ thus giving us an estimated relative error of about 16%. In reality, computations always counted errors in complete blocks. Looping continued until the 40-error limit was either attained or exceeded. In addition, we actually counted symbol errors and always assumed each such error was due to a single wrong bit. For low BER these approximations ought to be fairly accurate, while high error ratios are, first, not very interesting and, second, of admittedly dubious accuracy.

4. PROPERTIES OF OFDM

We turn now to some of the properties of OFDM that can be discovered through our simulations. First, note the amplitudes of the signals in Figure 2. In the rather nominal units used there, the rms value of the complex signal is 0.16. Thus, the individual components should have amplitudes of about 0.11, and indeed the signals do seem to hover about such a value. But at 6 μs (and again at about 44 μs*)* the quadrature component becomes three times as great. Such behavior is to be expected in an OFDM signal—if the coefficients in (1) have a random nature then at anyone time the sum will, by the central limit theorem, be approximately a gaussian random variable. This result provides one argument against the use of OFDM, since the transmitter will have to be able to handle these occasionally large voltages.

In what follows now we shall fix many of the system parameters to fit how we imagine HDTV will appear. We suppose B = 6 MHz and we choose N = 384—which is a nice round number (being 3/4 of 512), making the time period for a block T = 64 μs. This is almost the same as the 63.56 μs that is used for a horizontal line in today's NTSC television. The frequency spacing between carriers will be 15.6 kHz and we would expect the transformation between symbol blocks and the time domain to involve 512-coefficient FFT's. This means that 128 of the coefficients will be zeroed out. And for an even more realistic system we set a few more at each end of the bandwidth equal to zero so as to provide guard bands for adjacent channels. Keeping the other parameters fixed, we take off 200 kHz from each end, winding up with 360 carriers and a *used* bandwidth of 5.6 MHz. The results are illustrated in Figure 7 (the time domain signal) and in Figure 8 (the received sequence of symbols). We have used a simple BPSK modulation but have added in white noise with a 10 dB signal-to-noise ratio (SNR).

In Figure 9 we have plotted BER's *versus* SNR. The system is as just described, and for the modulation we have used both QPSK and DQPSK (*differential* QPSK). We have also plotted the well-known error curve for single-carrier QPSK. Note that our computed points match up very well with this curve, a fact that should not have surprised us. Under additive white noise there is no advantage in error rates for either single- or multi-carrier architectures. With only additive white noise there is also no advantage in DQPSK—indeed the difference in phase has twice the variance of the phase itself, and the error rate will be multiplied by two since each error in decoding affects the two adjacent symbols.

In these calculations of BER we consider an error ratio of 10^{-3} or 10^{-4} to be a near perfect, errorless result. This is because we assume the system is actually using COFDM

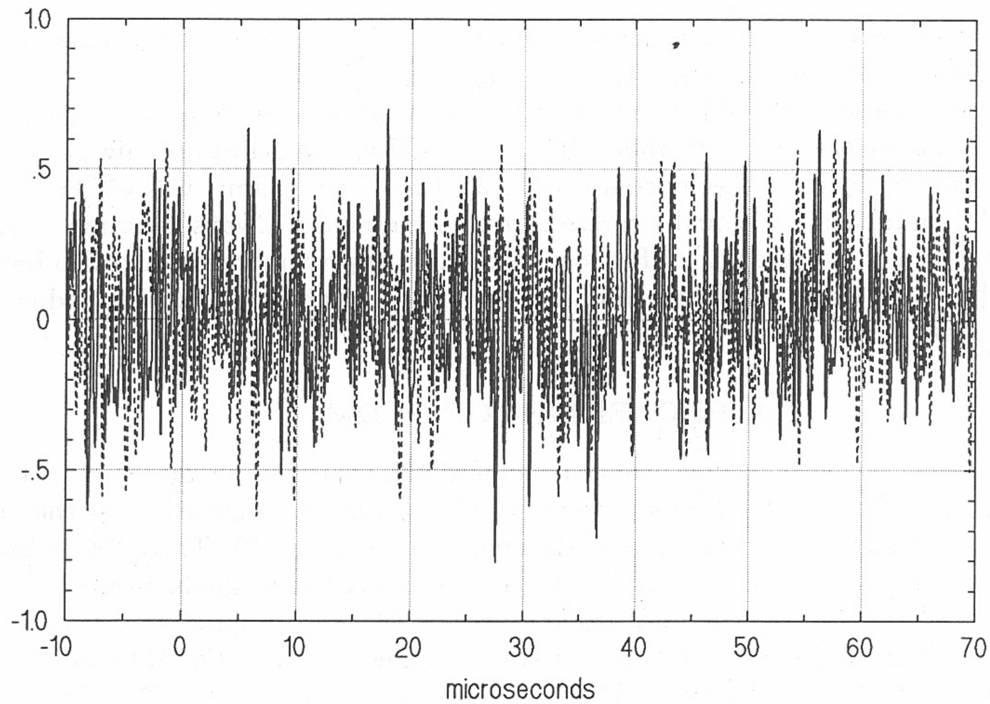

Figure 7. The complex signal from a simulation of HDTV using 360-carrier OFDM and BPSK modulation. One of the blocks extends from 0 to 64 μs. White noise is present at an SNR of 10 dB.

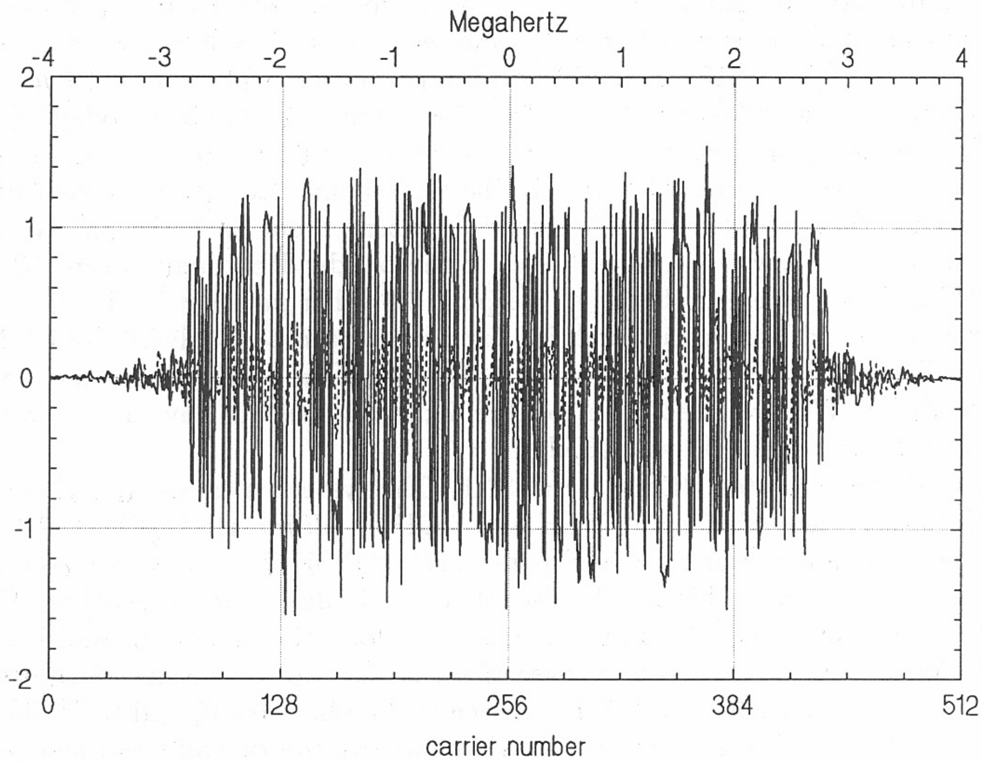

Figure 8. The received symbols from the block illustrated in Figure 7.

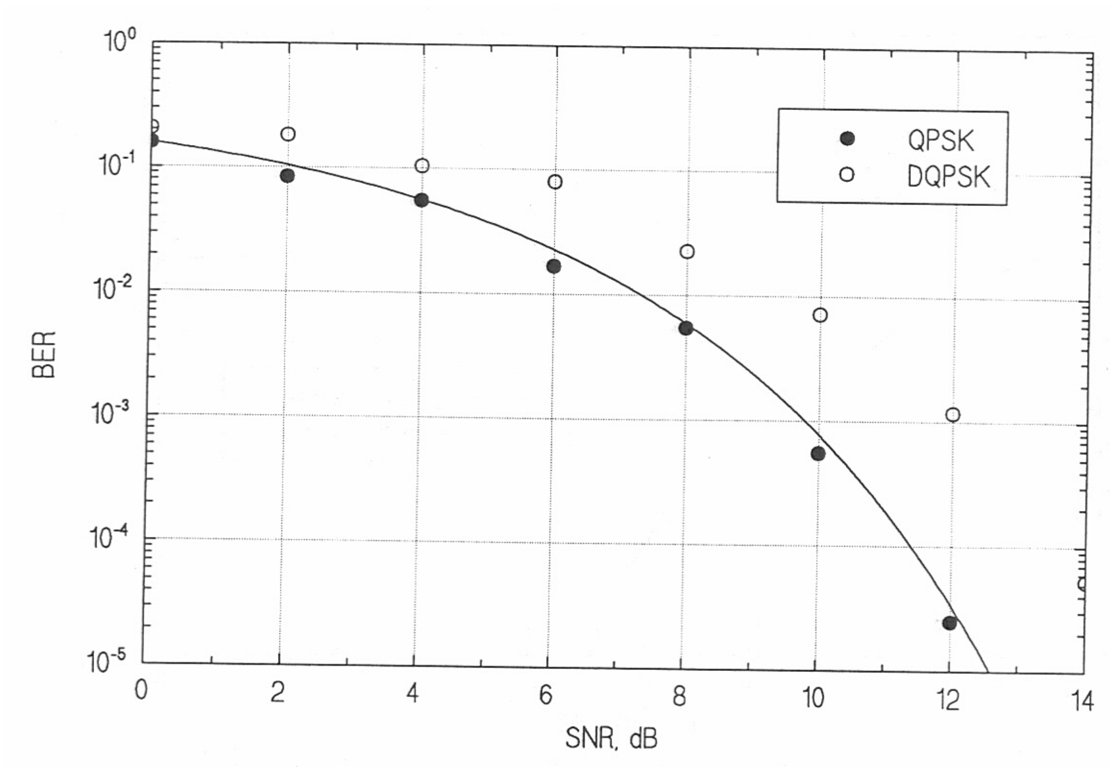

Figure 9. Bit error ratio *versus* signal-to-noise ratio for a simulated HDTV signal using OFDM and both QPSK and DQPSK modulation. The curve is the theoretical error curve for single-carrier QPSK modulation.

and that once the error ratio is this low the external block code will correct all remaining errors. This, of course, takes away from the modulation efficiency; for example, in the systems portrayed in Figure 9, the throughput will be 11.2 Mbit/s, but this rate includes whatever parity bits are necessary.

One advantage of OFDM is in the ease with which its spectrum can be shaped. This is especially valuable when interference problems arise, for in effect we can construct rejection filters with very precise rejection bands and with very sharp boundaries. We have already seen how guard bands can be introduced simply by deciding not to use some of the outside carriers, and the same principle can be used for any of the carriers. Thus if there is a cw (continuous wave) interferer one can avoid using the one or two carriers of the same frequency. Conversely, to prevent interfering with that other service, one sends a zero signal on those carriers. More to the point, present plans call for HDTV signals to share the spectrum with normal NTSC television, and one consideration will be how they interfere with each other. To combat such interference we have devised a “mask” that deletes from the OFDM signal frequencies surrounding the NTSC carrier, its chrominance subcarrier, and its audio carrier. In Figure 10 we have plotted BER *versus* the desired-to-undesired (D / U) ratio, both with and without that mask. The improvement is striking. In the other direction, one usually assumes that a high rate digital signal will look mostly like white noise, thus making that analysis one that has probably already been done. In this situation, however, it is the areas we have masked out where an NTSC signal would be particularly susceptible, and it is not hard to believe this scheme would help both ways.

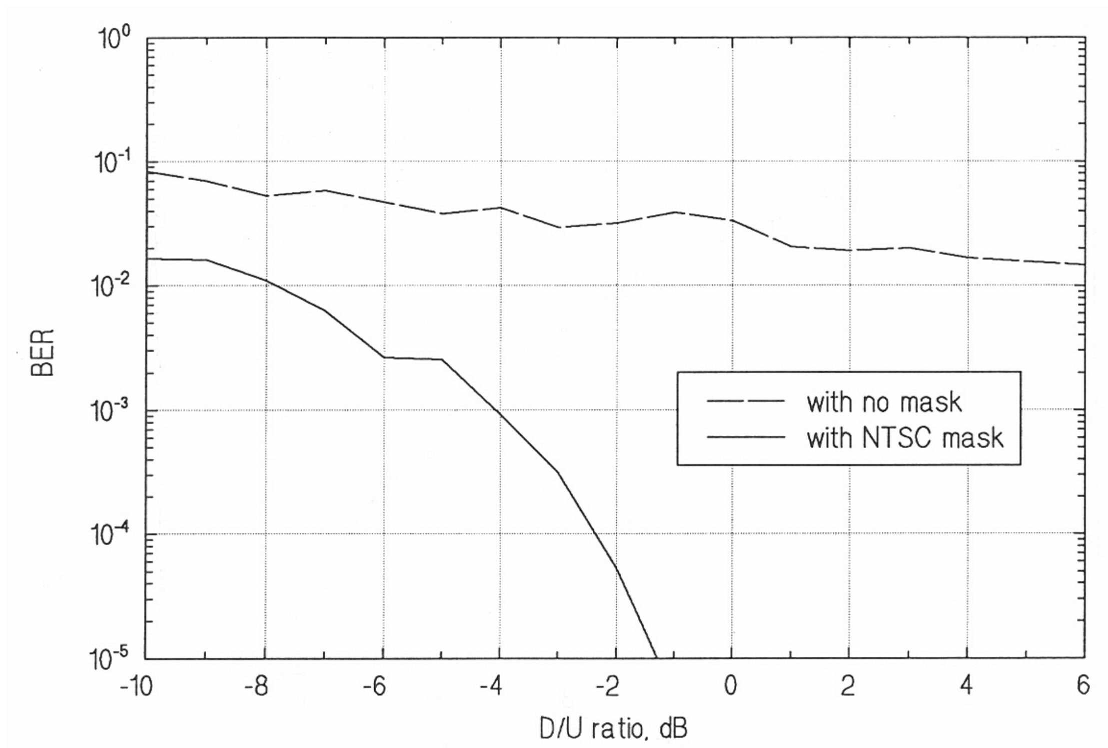

Figure 10. Bit error ratio *versus* desired-to-undesired ratio when an HDTV signal using OFDM suffers interference from the NTSC television signal portrayed in Figures 3 and 4. Results are shown with and without a mask that suppresses the stronger parts of the NTSC signal.

Of course, such a mask again reduces modulation efficiency—essentially we have removed a part of the spectrum from our use. As designed here the mask has left remaining 291 of the 384 carriers, and therefore the effective spectrum has been reduced to 4.5 MHz.

The principle feature that proponents stress is the ability of OFDM to overcome multipath because the signals change so slowly. More exactly, to avoid intersymbol interference one introduces into the signal a *guard interval* at the beginning of each block. Within this interval the previous block might degrade values of the current block, but at the receiver the interval and such degraded values will just be ignored. This means that the value of T in (1) is made deliberately large, $T = 1 / \beta + T_g$, where the interval T_g is enough to encompass all major multipath components. The fact that the block rate is slow means that the interval T_g will not take up relatively much time. To show how this behaves, in Figure 11 we have plotted the BER *versus* SNR for a QPSK and a DQPSK system when, in addition to the white noise, the signal has passed through a two-ray multipath channel. The figure compares results both without and with a 2-µs guard interval. From the figure, one notes that multipath does indeed wreck havoc on reception, that the use of a guard interval has improved the behavior, and that DQPSK shows now why it is sometimes the preferred modulation. (In fact, the DQPSK curve involving the guard interval is almost identical with the values in Figure 9 where there was no multipath.)

Of course, multipath also implies frequency selective fading and this brings up new problems for OFDM that the plot in Figure 12 illustrates. This graph shows the received

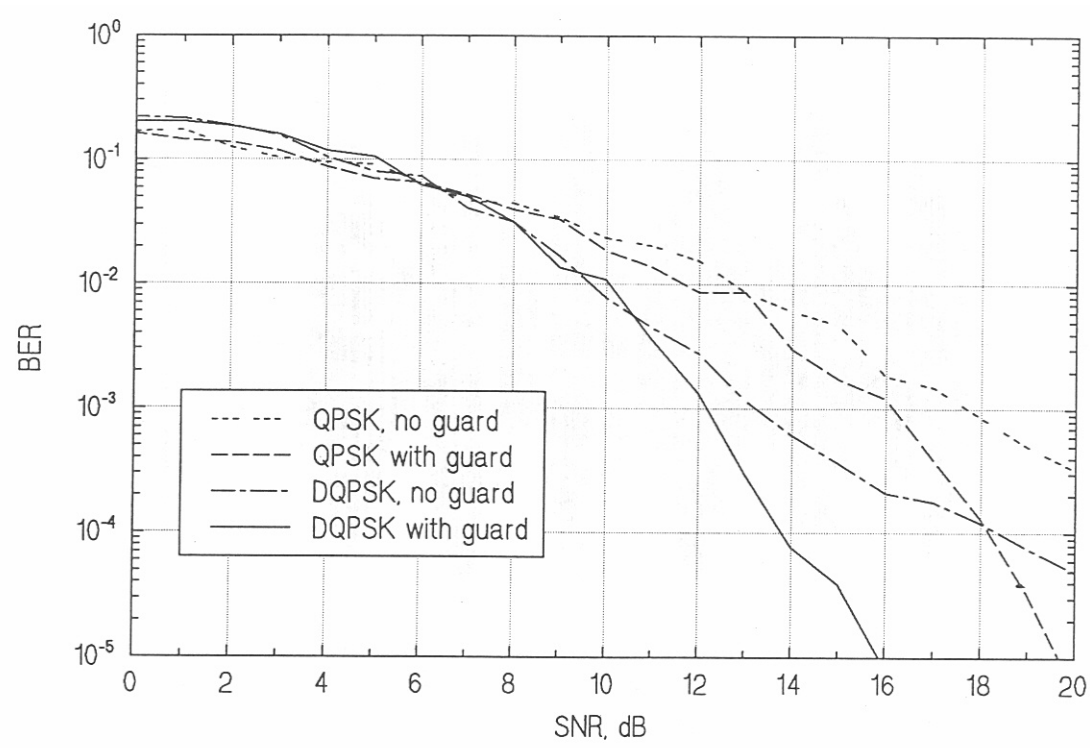

Figure 11. Bit error ratio *versus* signal-to-noise ratio when the channel also suffers from two-ray multipath with a 1-μs delay spread. Results are shown with and without an intersymbol guard interval of 2 μs.

sequence of symbols when BPSK modulation is used and the channel is subject to the same two-ray multipath used in Figure 11. Figure 13 shows the results with a more realistic channel having a tail of uncorrelated scatterers. Both figures portray rather dramatically the fading that occurs. Both amplitude and phase of the received signal are highly affected and that, of course, is why phase modulation is degraded and why DQPSK has the better performance.

But there will be even more trouble if amplitude modulation is involved. As we have said, HDTV will probably require symbols of 4 or 5 bits and so modulation cannot stop at QPSK but must go on, perhaps to ASK (amplitude-shift keying) or QAM. Clearly in such a situation the system will require an equalizer—or, as it might be phrased, each of the many carriers will require an AGC (automatic gain control). To effect such an equalizer, we have chosen a very simple process. We have argued that television will already require frame synchronization and so every so often we introduce an entire block of a known signal. This is read at the receiver and the (complex) ratio between what is received and what should have been received serves as our equalizer. For subsequent blocks, then, there are no convolutional computations, but only a series of multiplications. It sounds too perfect, although one should remember that the process of estimating AGC multipliers is also subject to noise and intersymbol interference.

The consequences of equalization are illustrated in Figure 14. There we show BER's for QPSK modulation when the transmission channel has the same two-ray multipath used previously. The curve without equalization is copied from Figure 11. The curve with

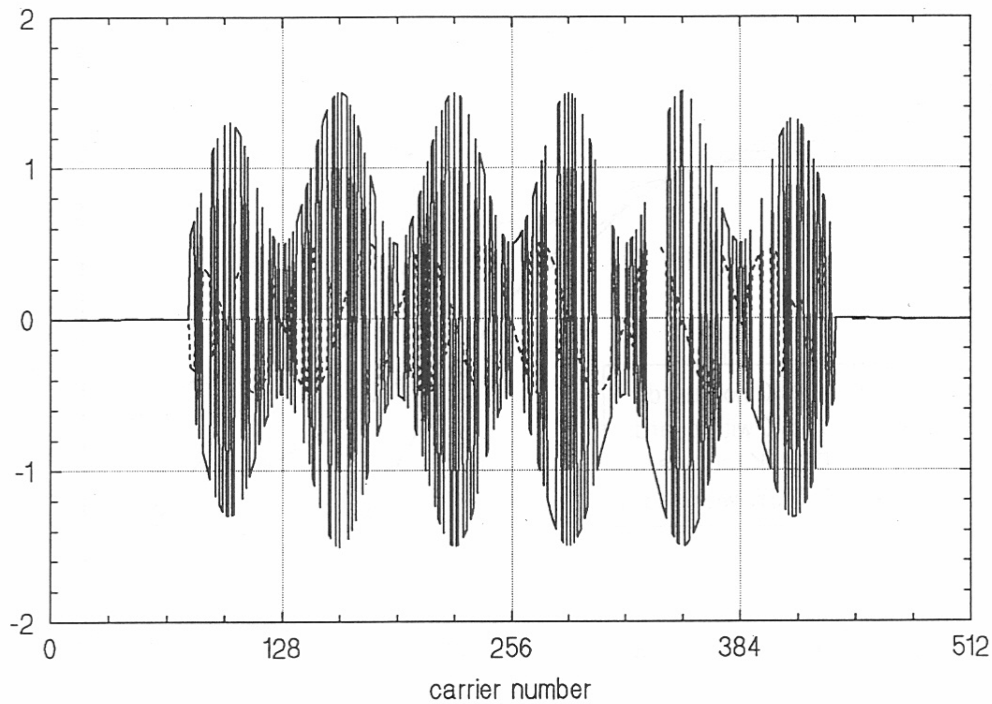

Figure 12. The received symbols from a system using BPSK. There is no additive noise, but the channel suffers from two-ray multipath with 1-μs spread.

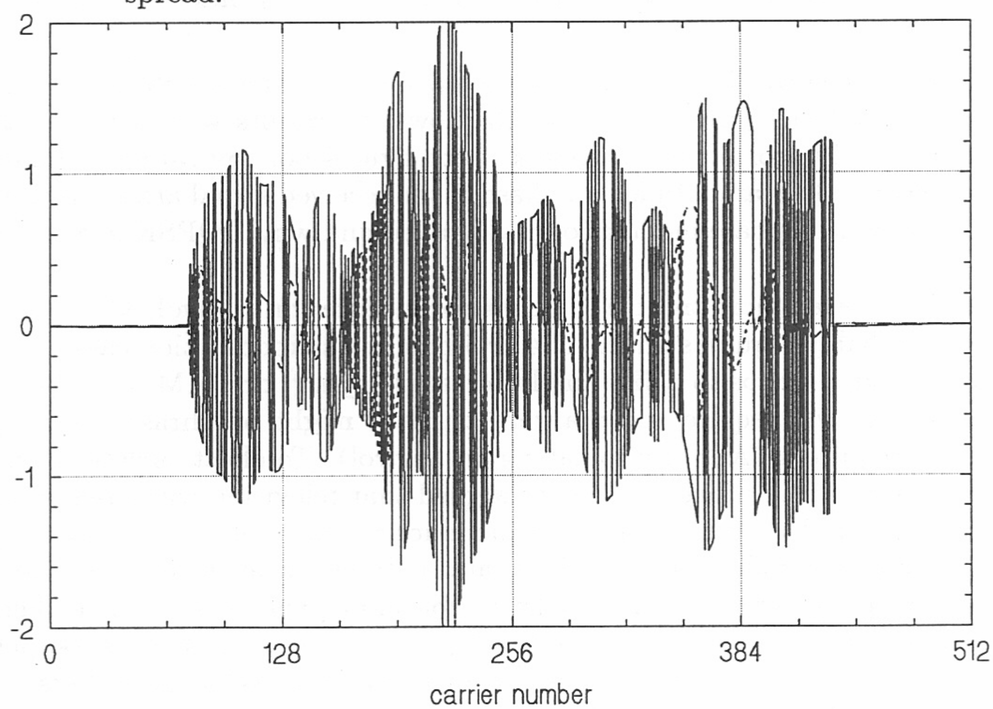

Figure 13. The received symbols from the same system illustrated in Figure 12. This time the channel has a tail of uncorrelated scatterers as in Figure 5.

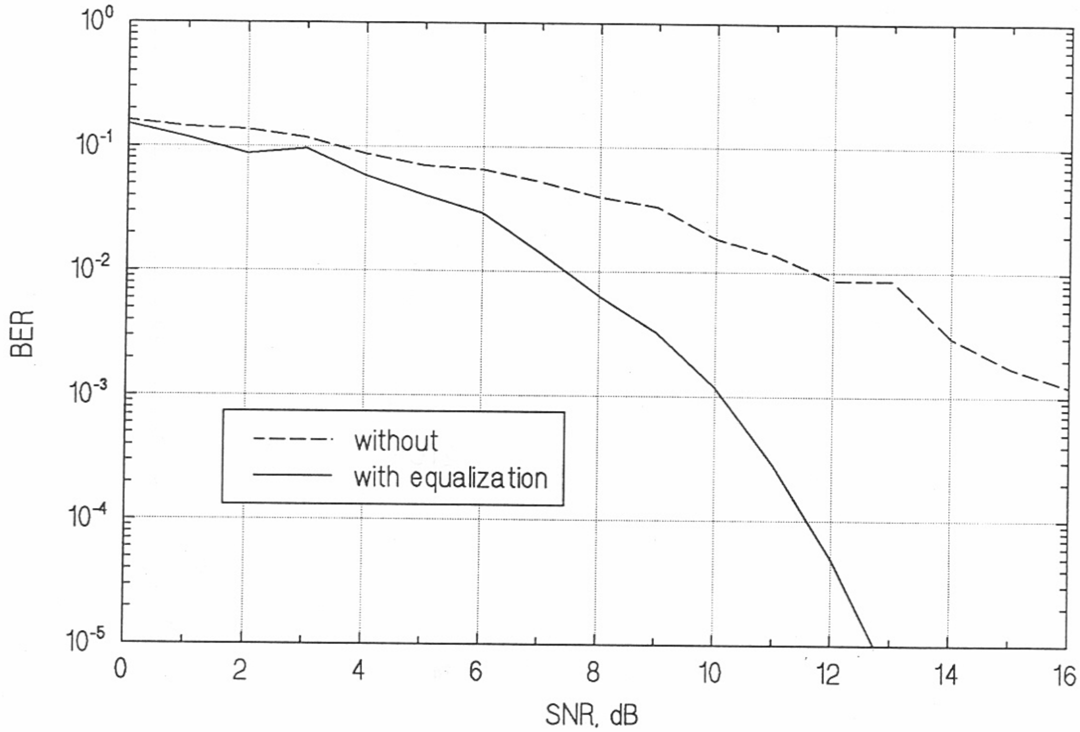

Figure 14. Bit error ratio *versus* signal-to-noise ratio showing the results of equalization. The modulation is QPSK and there is a guard interval. The channel suffers two-ray multipath.

equalization has values only a little worse than those of Figure 9.

If we are to be more realistic, we must at this point try to increase the throughput. To get to 4 bits per symbol we probably would use either 16QAM or 32TCM, and in Figure 15 we have plotted the usual comparison between the two when only white noise is involved. Just as expected, the error-correction ability of the TCM provides a 3-dB advantage despite its tighter constellation.

Finally, in Figure 16 we have plotted BER *versus* SNR for a system using 32TCM and passing through a channel with multipath of varying delay spreads. The system includes an equalizer and a 2-μs guard interval. The multipath simulation has a tail of uncorrelated scatterers and the power in the tail is 6 dB below that in the direct path. The 2 μs for the guard interval is probably shorter than a real service would require, as the results in Figure 16 certainly indicate. Looking at such BER's we should recall that according to Shannon's theorem we obtain error-free throughput of 20 Mbits/s through a 6-MHz channel only if we have a signal-to-noise ratio of at least 9.6 dB.

5. CONCLUDING REMARKS

Generally speaking, our studies here have confirmed many of the claims made for the OFDM architecture. These claims are not really that the scheme gives better results, but that these results are easier to obtain and therefore probably more robust. The ease with which an equalizer can be designed is one example. But perhaps most interesting is the ease with which one can shape the spectrum. We have chosen here to put guard bands at

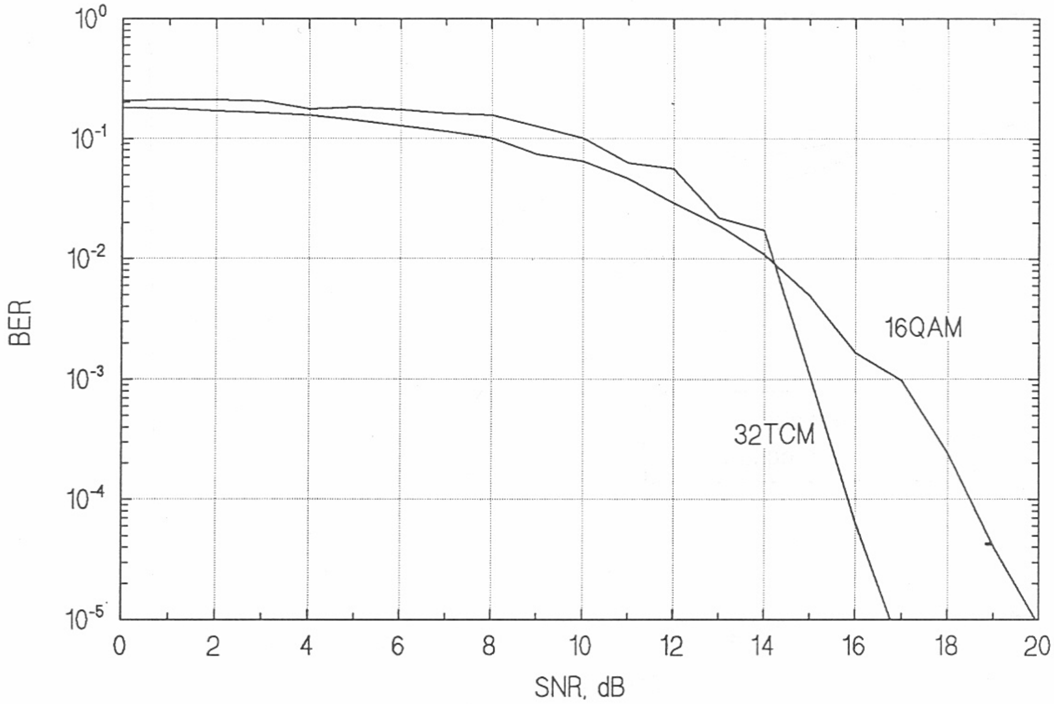

Figure 15. Bit error ratio *versus* signal-to-noise ratio showing a comparison between 16QAM and 32TCM. Both systems are equalized and there is no multipath.

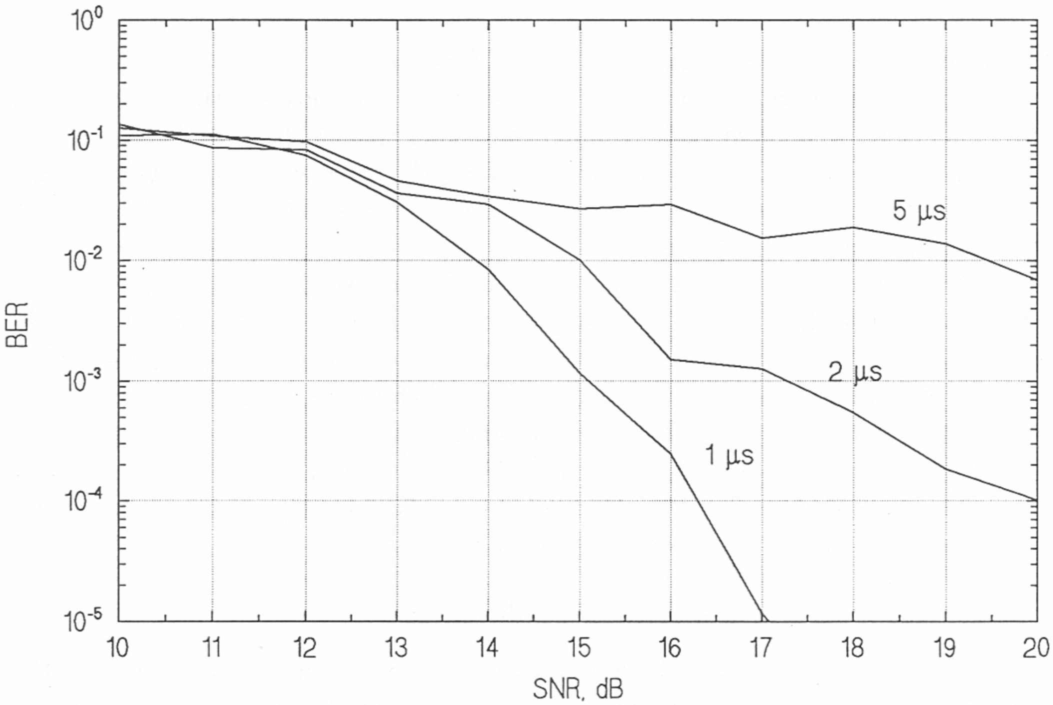

Figure 16. Bit error ratio *versus* signal-to-noise ratio for a system using 32TCM. Multipath (with a tail of uncorrelated scatterers) is present with varying delay spreads as indicated.

the two ends of our 6-MHz channels—we did this to be conservative, but it is not obvious that it was necessary. And the BER's in Figure 10 (far better than those for white gaussian noise) seem to indicate that we removed more carriers than was necessary.

We have shown how a guard interval along with a simple equalization module will go a long way towards eliminating the effects of multipath. The same sort of reasoning suggests a solution to the problems of *simulcasting* in which more than one station broadcast the same program in the same channel to overlapping areas. If it works there are several places where the technique could be used to advantage. We could replace one large transmitter with several small ones, thus providing a more uniform signal over the service area, and a signal that drops off more quickly outside the service area. This latter means the channel can be reused more quickly, thus increasing "spectrum efficiency" (as defined, *e.g.,* in [R. J. Matheson, "A survey of relative spectrum efficiency of mobile voice communication systems," to be published]). For HDTV it might allow a service area closer to a co-channel NTSC service area than would otherwise be the case. This use of simulcasting could be extended to a more careful shaping of service areas or it might simply be used as a repeater to illuminate a shadowed area.

The problem, of course, is that self interference can produce undesired, even disastrous, effects. On the other hand, if one thinks of two transmitters as producing two rays of the same signal, then one can say it looks just like multipath propagation. Since OFDM solves multipath problems it should also manage simulcasting. And indeed it might, although there are still questions. First, there would need to be guard intervals at *both* ends of the time interval T. The signals would have to be carefully timed so that they leave all transmitters almost simultaneously. And the signals would have to be themselves almost identical. A slight error in center frequencies, for example, would cause the ripple, as seen in Figure 12, to move steadily one way or the other, rendering our simple equalizer ineffective.

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"Orthogonal Frequency Division Multiplexing" is an interesting system architecture for high data-rate digital transmission that has been receiving renewed attention. This report is a discussion of how it works and of how it might be used to carry High Definition Television. The principle tool is a computer simulation that allows us to look at many aspects of an OFDM system. It is concluded that the claims for the system largely hold up, and that its use might simplify many technical problems for HDTV.

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