**Project: OFDM Technology**

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| **Introduction**  In modern communication system, the bandwidth is getting larger and larger and the transmission rate getting faster and faster. This leads to a frequency selective channel, where multiple paths in the propagation environment create distortions in the transmitted. For example, if there are two propagation paths, the receiver may observe the signal:    Besides frequency selective fading, wideband channel will also lead to inter-symbol interference (ISI).  There are three ways to achieve delivering signals without ISI:   * Approach 1: Increase the time interval to make it larger than the delay extension of multipath effect. But it will reduce data transfer rate. * Approach 2: Channel equalizer.     But as the maximum delay increases, the complexity of equalizers is correspondingly more and more complicated, which leads to the high computational complexity and unrealizable.   * Approach 3: Multi-carrier modulation. In this project, we discuss the OFDM technology. Orthogonal Frequency Division Multiplexing (OFDM) is a digital multi-carrier modulation technique extending the concept of single subcarrier modulation by using multiple sub-carriers over the channel.   There are some advantages of OFDM technology:   * Spectra efficiency; * Against frequency selective fading; * The ISI can be completely eliminated through the use of a cyclic prefix; * Use multi-subcarriers to decrease the complexity of equalizer.   And there are also some applications of OFDM technology:   * 4G mobile communication system; * Multi Input Multi Output (MIMO) technology; * Ultra Wide Band (UWB) radios; * IEEE 802.11a Wireless LAN; * Digital Audio Broadcasting (DAB); * HiperLAN/2 (High Performance Radio LAN).   **Theoretical analysis of OFDM**   1. **Problems caused by multipath channels in wideband transmission**     As we all know, when the signal is propagating in the wireless channel, the received signal is the result of a superposition of multiple channels due to the reflection, refraction and so on. And there are attenuation and time delay in other channels. The time delay of different paths is superimposed together to form the “Multipath Effect”. The impulse responses of all of the paths can be considered as:  So the total impulse response of multipath is:    When the time delay is serious, narrow band channel will become wideband channel. According to the impulse response of multipath, the chance of inter symbol interference will greatly increase with the bandwidth of the channel increasing, which resulting in serious distortion of the received signal. Therefore, how to recover the transmitted signal in the received signal is a difficult problem needed to be solved.     1. **Frequency selective fading**   Another serious problem caused by the multipath effect is frequency selective fading. Mentioned in the last part, the total impulse response of multipath is:  According to DTFT, the frequency response of is:  The frequency response of is:  So, the frequency response of multipath is:  The magnitude of the frequency response of multipath is as follows:    From the picture, what is obvious is that the magnitude response is 0 at some certain frequency while the magnitude response is greater than 0 at other certain frequency, which is called frequency selective fading. OFDM can resist frequency selective fading.    The whole channel can be divided into several sub-channels. Each of sub-channel can be considered as experiencing the flat fading rather than frequency selective fading. OFDM can  assign subcarriers to those sub-channels. Each sub-channel can overlap as long as the sampling point on the subcarrier is the zero crossing of the other subcarriers.   1. **Basic idea of OFDM technology**   First of all, we should know the basic idea of multicarrier modulation. The basic idea of multicarrier modulation is to divide the transmitted bitstream into many different substreams and send these over many different subchannels. Typically, the subchannels are orthogonal under ideal propagation conditions. The data rate on each of the subchannels is much less than the total data rate, and the corresponding subchannel bandwidth is much less than the total system bandwidth. The number of substreams is chosen to ensure that each subchannel has a bandwidth less than the coherence bandwidth of the channel, so the subchannels experience relatively flat fading. Thus, the intersymbol interference on each subchannel is small.  In the discrete implementation of multicarrier modulation, called orthogonal frequency division multiplexing (OFDM), the ISI can be completely eliminated through the use of a cyclic prefix.  And there are some basic ideas of OFDM technology in the following:   * Use orthogonal subcarriers, which leads to better bandwidth efficiency compared with conventional FDM.     Orthogonal subcarriers   * Divide channel into several subchannels through subcarriers so that each subchannel could be considered as a flat channel. This indicates that why OFDM could be resistive to multipath fading. * A guard interval is added to each symbol to minimize the channel delay spread and intersymbol interference (ISI) and intercarrier inference (ICI). Moreover, cyclic prefix is proposed to play both roles of guard interval and linear-cyclic convolution conversions. * Use efficient FFT algorithm and serial to parallel conversion to ensure a high data rate and low computational complexity for DFT. Some related works show that FFT algorithm uses divide and conquer method to minimize computational complexity to which is much more efficient than direct DFT calculation method (with a computational complexity . * Inserting null tones (DC subcarriers and padding zeros) could not only help the FFT/IFFT compute faster but also contribute to estimating the discrete time signal closer to continuous time signals. * Simpler channel equalization: An advantage of OFDM is that using multiple subchannels, the channel equalization becomes much simpler. This means that OFDM could guarantee a high data rate and meanwhile a simplicity of designs of channel equalizer and frequency correction. * OFDM symbol format considered in this lab is shown below. Each subcarriers stands for an OFDM symbol which is consisted of CP and transmitted bits. All the OFDM symbols use same training sequence.     OFDM symbol format  The basic OFDM diagram is shown below:     1. **IFFT and FFT**   It is important to keep in mind at the outset that the FFT is not a new transform. It is simply a very efficient way to compute an existing transform, namely the DFT. As we saw, a straightforward implementation of the DFT can be computationally expensive because the number of multiplies grows as the square of the input length. The FFT reduces this computation using two simple but important concepts. The first concept, known as divide-and-conquer, splits the problem into two smaller problems. The second concept, known as recursion, applies this divide-and-conquer method repeatedly until the problem is solved. Consider the defining equation for the DFT and assume that N is even, so that N/2 is an integer:  Suppose we break the sum into two sums, one containing all the terms for which n is even, and one containing all terms for which n is odd:    Then we do the following trick:  First, we define two new N/2 point data sequences, which contain the even and odd numbered data points from the N point sequence:  This separation of even and odd points is called decimation in time. So:  These two facts may be combined to yield a simpler expression for the N point DFT:  So, we can get a simple figure about the procedure on the FFT:    If we try to use the recursion method to implement the FFT, then we get:    Finally, we have a faster way of calculating the DFT and IDFT, whose time cost is lower than NlogN.   1. **Cyclic prefixes and cyclic convolution** 2. **Cyclic prefixes**   Consider a channel input sequence x[n] = x[0], ..., x[N − 1] of length N and a discrete time channel with finite impulse response (FIR) h[n] = h[0], ..., h[µ] of length µ + 1 = Tm/Ts, where Tm is the channel delay spread and Ts the sampling time associated with the discrete time sequence. The cyclic prefix for x[n] is defined as {x[N − µ], ..., x[N − 1]}: it consists of the last µ values of the x[n] sequence. For each input sequence of length N, these last µ samples are appended to the beginning of the sequence. This yields a new sequence x˜[n], −µ ≤ n ≤ N − 1, of length N + µ, where x˜[−µ], ..., x˜[N − 1] = x[N − µ], ..., x[N − 1], x[0], ..., x[N − 1]. Note that with this definition, x˜[n] = x[n]N for −µ ≤ n ≤ N − 1, which implies that x˜[n − k] = x[n − k]N for −µ ≤ n − k ≤ N −1    Suppose x˜[n] is input to a discrete-time channel with impulse response h[n]. The channel output y[n], 0 ≤ n ≤ N − 1, is then  Taking the DFT of the channel output in the absense of noise then yields  Y [i] = DFT{y[n] = x[n]h[n]} = X[i] H [i], 0 ≤ i ≤ N − 1  The input sequence x[n], 0 ≤ n ≤ N −1, can be recovered from the channel output y[n], 0 ≤ n ≤ N − 1, for known h[n] by  Note that y[n], −µ ≤ n ≤ N −1, has length N + µ, yet from (12.20) the first µ samples y[−µ], ..., y[−1] are not needed to recover x[n], 0 ≤ n ≤ N − 1, owing to the redundancy associated with the cyclic prefix.   1. **Cyclic convolution**   It is a calculation that similar to the linear convolution, but has some slight differences. Consider two sequence that length is N, which is g[n] and h[n], defined on the interval . Those linear convolution results length is 2N-1,  That is:  Two sequences with length N have been expanded to 2N-1 by zeroing. The longer sequence y[n] comes from the time reversal of the shorter sequence h[n] and the linear shift to the right after the reversal.  In order to establish an operation similar to convolution to generate sequences of length N, we first use the circular time reversal operation, and then apply the circular time shift. Such operation is called circular convolution.  It is called the N point circular convolution. Noted as:    It also has the matrix form, that is:     1. **Subcarrier and null tone mapping**   Because the number of points for FFT and IFFT is usually , for example, in 4G LTE, we assume that there are 1200 symbols, but in this case, we need to map them on 2048 subcarriers, which is subcarriers.  To map the symbols to all the subcarriers, we need to insert DC component and null tone to these symbols. For DC component, we usually insert it in the first position. And for null tone, for example, there we have 15 symbols, and we want to map them on 32 subcarriers, so we must have 17 zero paddings in all these 32 symbols. Except for the DC component, the remaining 16 zero paddings are all the null tones, and we insert them into the middle of the FFT.    The case of mapping 6 symbols to 8, i.e., subcarriers is shown below:    **Lab results & Analysis**   1. **Simulation of OFDM modulation and demodulation** 2. **Block diagram**  * **Modulation**     The figure shown above is the block diagram of the OFDM modulator.   * **Demodulation** * **Demodulator**      * **FEQ**      1. **Program process**  * **Modulation**     For OFDM modulator, after QAM modulation, we first convert serial input stream to parallel stream, whose size is N-K, then insert K null tones. In the third step, we need to carry out N points IFFT to get the waveform from the Fourier series. And then, add cycle prefix to it. Finally, convert parallel stream to serial stream, then pass the output into the wireless channel.   * **Demodulation**     For OFDM demodulator, the first thing to do is to convert serial input stream to parallel stream, whose size is . Then, remove the cycle prefix. The third step is to perform FFT. After that, apply frequency domain equalizer. Immediately after that, remove null tones. Finally convert parallel stream to serial stream and perform QAM demodulation.   1. **Simulation result**     The figure shown above is the simulation result of the OFDM modulator and OFDM demodulator. It is obvious that the simulation result is perfect, which means the OFDM modulator and OFDM demodulator is correct.   1. **Result of frequency selective fading channel**   Task1_800kHz Task1_1MHzTask1_400kHz  Sample Rate = 400KHz Sample Rate = 800KHz Sample Rate = 1MHz  Task1_5MHzTask1_4MHzTask1_2MHzTask1_1MHz  Sample Rate = 2MHz Sample Rate = 4MHz Sample Rate = 5MHz  From the result, we can find that with Sample Rate increases: the shape of the frequency response changes from Narrow band to the Wide band and the fading changes from Flat fading to the Frequency selective fading.   1. **Result of frequency offset sensitivity of OFDM technology**   Under [10 4 10 4 ] N=64  Task2_200HzTask2_30Hz  Task2_10Hz  Frequency Offset = 10Hz Frequency Offset = 30Hz Frequency Offset = 200Hz  Under [20 20 4 4 ] N=64  Task2(20,20,4,4)_64_100HzTask2(20,20,4,4)_64_50HzTask2(20,20,4,4)_64_200Hz  Frequency Offset = 10Hz Frequency Offset = 100Hz Frequency Offset = 150Hz  Under [10 4 10 4 ] N=512  Task2_512_50HzTask2_512_150HzTask2_512_100Hz  Frequency Offset = 50Hz Frequency Offset = 100Hz Frequency Offset = 150Hz  Task2_1024_150HzUnder [10 4 10 4 ] N=1024  Task2_1024_100HzTask2_1024_50Hz  Frequency Offset = 50Hz Frequency Offset = 100Hz Frequency Offset = 150Hz  Under [20 20 4 4 ] N=1024  Task2(20,20,4,4)_1024_200HzTask2(20,20,4,4)_1024_150Hz  Task2(20,20,4,4)_1024_50Hz  Frequency Offset = 50Hz Frequency Offset = 100Hz Frequency Offset = 150Hz  From the result, we can find that with the number of subcarriers increases, the recovery quality becomes worse. Later, we will show our USRP verification.   1. **Result of the effect of the number of subcarriers on the system**   In this section, the effect of the number of subcarriers is analyzed.  Under [10 4 10 4] frequency offset=50 Hz    N=512 N=1024  Under [10 4 10 4] frequency offset=100 Hz    N=512 N=1024  Under [10 4 10 4] frequency offset=150 Hz    N=512 N=1024  From the above pictures, a conclusion can be gotten: as the number of subcarriers increasing, the quality of recovering become worse and worse when all other conditions and parameters are fixed. So, in order to improve the quality of recovering, other conditions and parameters should be changed.  Under [20 20 4 4] N=1024    Under [4 4 4 4] N=1024    Comparing the quality of recovering of [10 4 10 4] N=1024, [20 20 4 4] N=1024, [4 4 4 4] N=1024 at the same frequency offset, what can be found is that reducing oversample factor and sample rate of TX and RX can improve the quality of recovering when the number of subcarriers increasing.   1. **USRP verification**   In this part, we use USRP to verify the frequency selectivity of wireless channel and sensitivity to frequency offset.   * **Frequency selectivity of wireless channel**       From the result, what can be found is that with the sample rate increasing, the shape of the frequency response changes from the narrow band to the wide band and the fading changed from flat fading to frequency selective fading. The results are in agreement with the theoretical analysis and simulation results.   * **Sensitivity to frequency offset**   Under [10 4 10 4] N=64    Under [10 4 10 4] N=512    Under [10 4 10 4] N=1024    From the result, what can be found is that with the frequency offset increasing, the quality of recovering become worse and worse when all other conditions and parameters are fixed. What’s more, with the number of subcarriers increasing, the quality of recovering also become worse and worse. The results are in agreement with the theoretical analysis and simulation results.   1. **Result of the high-order modulation of subcarrier** 2. **Modulation**  * **16QAM**     For 16QAM modulation, we first extract the bits in the bitstream into groups of every 4 bits, and then convert them to decimal numbers and map them to the corresponding 16QAM symbols as index values. In another words, we map 4 bits into a symbol, for example, bit stream 10110111, we divide it into 1011 and 0111, and from the process of converting binary to decimal, we know that the highest bit in the case has a weight of 8, and the second has a weight of 4, and the third and the lowest has a weight of 2, and 1, respectively, and then, we use the decimal index we get to find the corresponding complex number, i.e., symbol in the 16-QAM symbol mapping table to implement the process of mapping the bit stream to symbol.   * **64QAM**     For 64QAM modulation, its implement is similar to that of 16-QAM modulation. Specifically, we need to notice that in 64-QAM modulation strategy, we map every 6 bits into a symbol, and the highest bit in the case has a weight of 32.   1. **Demodulation**  * **16QAM**     For 16QAM demodulation, first of all, we need to normalize the energy of the symbols in the symbol mapping table. Then differentiate received symbol r with reference symbols sm, and convert the result to the polar coordinate, take its modulus, which is the a series of 2 norm between the received symbol r and the reference symbol sm. And then find the index of the minimum element of the 2 norm array, where the index of the minimum element is the decimal number of the bits we want to recover. Outside the for loop, divide the decimal number by 2, divide the quotient by 2 again, and so on until the quotient is 0, and mark the remainder of each step, i.e., 0 or 1 next to it, and write it backwards to get the corresponding binary bits, which is the recovered bits we want to get.   * **64QAM**     For 64QAM demodulation, its implement is the same as that of 16-QAM demodulation, so I will not go into details here.   1. **Simulation result**  * **16QAM**     The figure shown above is the simulation constellation of 16QAM.   * **64QAM**     The figure shown above is the simulation constellation of 64QAM.  Note: In this part, we need to notice that we should increase the amount of the packet length and also the number of subcarriers.   1. **USRP verification**  * **16QAM**     The figure shown above is the constellation of 16QAM by using USRP to carry out verification. From the figure, we can find that our 16QAM modulation is successful.   * **64QAM**     The figure shown above is the constellation of 64QAM by using USRP to carry out verification. From the figure, we can find that our 64QAM modulation is successful.   1. **System implementation of image transmission** 2. **Image source**     At the transmitter, we replace the Dr. Wu’s source with the image source subVI, which is built by ourselves. In the subVI, we first read the image file through the path and convert it into a binary two-dimensional array, and then use **32bit\_to\_bitstreamV2.vi** to convert the two-dimensional array into a bitstream output. And proceed to the next step.   1. **Image recovery**     At the receiver, we use **bitstream\_to\_32bitv2.vi** to convert the recovered bitstream back to a two-dimensional array, namely the pixel graph, and finally draw an image to get our recovered image.   1. **Simulation result**  * **Transmitted image**     The figure shown above is the transmitted image for our simulation of the image transmission.   * **Recovered image**     The figure shown above is the recovered image for our simulation of the image transmission.   1. **USRP verification**     The figure shown above is the recovered image by using QPSK modulation strategy to carry out image transmission on USRP platform. From the figure, we can find that our image transmission is successful.    The figure shown above is the constellation of the received image by using QPSK modulation strategy to carry out image transmission on USRP platform. From the figure, we can find that our image transmission is successful. | |
| **Experience**  **Experience**  **岳翼遥:**   1. In this section, we have learned the principle and implementation method of fast Fourier transform. Fast Fourier transform is realized successfully. The advantage of the fast algorithm is found by comparing it with the real Fourier transform. And we clearly understand the divide and conquer algorithm programming idea, which will be a very important idea to speed up computer computing. In addition to this idea, it is followed by the idea of recursion. We decompose complex problems into small problems and recursively process multiple problems. Then we solve small problems and recursively synthesize them into large problems to complete the solution. This is a very important algorithm to reduce the computational complexity. As we all know, the computing resources and memory of computers are limited. People need to improve the computing power and robustness of the algorithm while reducing the computing cost to design an excellent algorithm. 2. Secondly, we learned the important use of cyclic convolution in ofdm, which is a very important method in communication systems and digital signal processing. We have learned that in modern communication systems, how to use the channel efficiently is a fundamental problem, and how to use the existing resources of the channel to allocate resources appropriately is our primary consideration.   **孙逸涵:**   1. There are always one bunch of constellation points converging at the origin. After observing the received waveform, we found it might be because of the wrong frame synchronization of the diagram. 2. When doing high order OFDM modulation, we need to increase the amount of the packet length and also the number of subcarriers. Otherwise, we cannot get a good received constellation with whole constellation points. 3. In the part of the image transmission, when we finish our image transmission program, and carry out simulation at the first time, we cannot simulate successfully. And after our debug, we find that we made an incredibly simple mistake, which is that we run our program in the incorrect place. Then we run our program in the OFDM\_simulator or the top\_ofdm\_tx and top\_ofdm\_rx again, it run successfully, which means our programming and simulation is correct. 4. In order to run the program properly on USRP platform, we should first to modify the hardware clock configuration from **MIMO** to **Internal**.   **张旭东:**   * + - 1. In the receiver, if we use the small USRP, we must do the following change or our program will make error:      * + - 1. Communication engineering technology always combined with mathematical theory, which will be helpful in solving important problems in the engineering       2. It is a good way to use simulation to help us understanding the principle. For example, in this lab, we theoretically analyze the frequency offset and shape of the frequency response according to the theoretical knowledge and do the USRP simulation to verify our analysis.       3. To run the image OFDM communication program, it should be noticed that the size of the picture should not be very large or the computer will broke sown.   **Group division**  **岳翼遥:** PPT making, report writing, USRP verification, basic tasks, debugging  **孙逸涵:** PPT making, report writing, advanced tasks, modulator, debugging  **张旭东:** PPT making, report writing, USRP verification, demodulator, debugging  **Photo of the presentation**    **In-class lab screenshot**  **岳翼遥:**      **孙逸涵:**          **张旭东:** | |
| **Score** | 100 |