Project：OFDM

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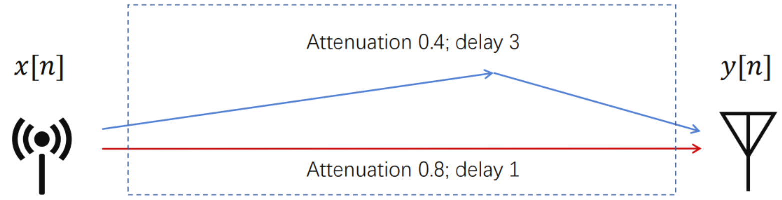
Introduction

1. The problems of multipath wideband transmission

Multipath interference comes from the multipath propagation of signals. Since wireless communication uses free space as the transmission medium, electromagnetic waves carrying information will reach the receiving end through multiple paths, thus bringing multipath interference. Multipath situation brings inter-symbol interference, which requires the channel equalization at the receiver.



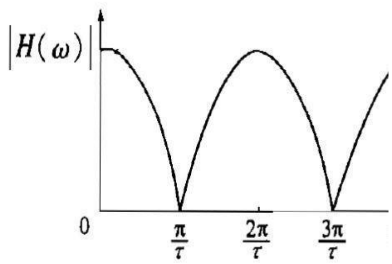
$$h_1[n] = 0.8\delta[n-1] + 0.4\delta[n-3] + \cdots$$

Assuming the same channel attenuation, only the delay is considered

$$R(\omega) = AF(\omega)e^{-jw\tau_0} + AF(\omega)e^{-jw(\tau_0+\tau)} = AF(\omega)e^{-jw\tau_0}(1+e^{-jw\tau})$$

Obtain the spectral function of the channel

$$H(\omega) = \frac{R(\omega)}{F(\omega)} = Ae^{-jw\tau_0}(1+e^{-jw\tau})$$



As for the effect of wideband channel, according to Nyquist's first criterion: the highest symbol rate of an ideal low-pass system equals 2 times the system bandwidth. In the digital communication system, the bandwidth is approximately equivalent to the transmission rate.

Then, if the signal bandwidth (transmission rate) is larger than a certain scale, we consider the system as a wideband system. Thus, the multipath interference will be generated. If the signal bandwidth is less than a certain scale, the system is defined as a narrowband system, without considering the influence of multipath interference. This "scale" is called the coherent bandwidth of the channel and is the reciprocal of the maximum delay difference generated by the signal propagating through multiple paths.

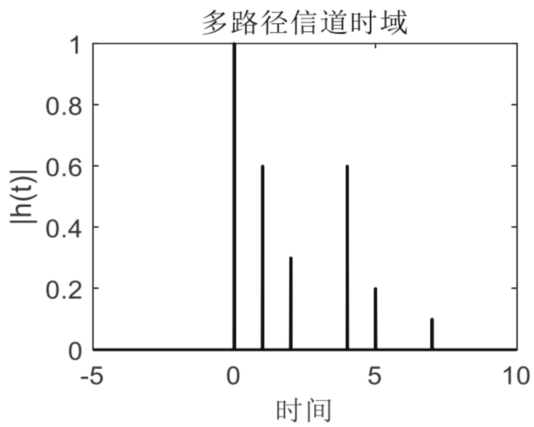
$$B_c = \frac{1}{\tau}$$

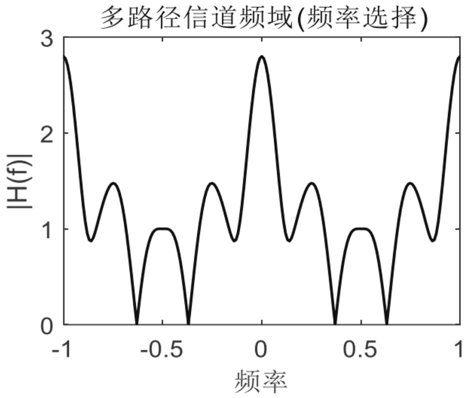
That is, the larger the maximum delay difference, the smaller the coherent bandwidth of the channel.

Modern transmission systems require faster transmission rates, that is, the bandwidth is larger. As for wideband transmission system, it has to face the problem of multipath interference. One solution to this challenge is using OFDM modulation method to decrease the transmission rate of each path.

2. Frequency selective fading channel

The periodic fading caused by multipath interference of the frequency response, calls frequency selective fading. For frequency selective fading the fading characteristics is different in different frequency bands.





The multipath effect of wireless channels results in frequency selectivity. For wireless channel, its complex spatial pattern formed the integrated magnetic wave propagation environment, the spatial pattern of have appropriate physical size and characteristics of propagation of electromagnetic waves of different frequencies is different. So as in which transmit magnetic wave frequency change, the channel response is also constantly change, which is called the essential reason of frequency selectivity.

Another cause of frequency selective fading is Doppler effect. The real physical channel is a time-varying channel. Which define the channel impulse response time interval of maintaining the "constant" statistical average channel coherence time T. If a symbol time length is shorter than the coherence time, then the entire symbol waveform can be spread during the transmission channel is "consistent" the spread of great distortion is not going to happen. Opposite to produce time selective fading.

In the case of frequency domain selection, the response in the frequency domain is different in different frequency domains. The way OFDM is transmitted and demodulated depending on orthogonality will have an impact. This will break the orthogonality between different subcarriers of OFDM during transmission, resulting in demodulation failure.

3. OFDM basic idea:

Orthogonal frequency division multiplexing (OFDM) is multicarrier modulation technique that applies the channel estimation, frequency offset estimation and linear equalization at receiver with low complexity.

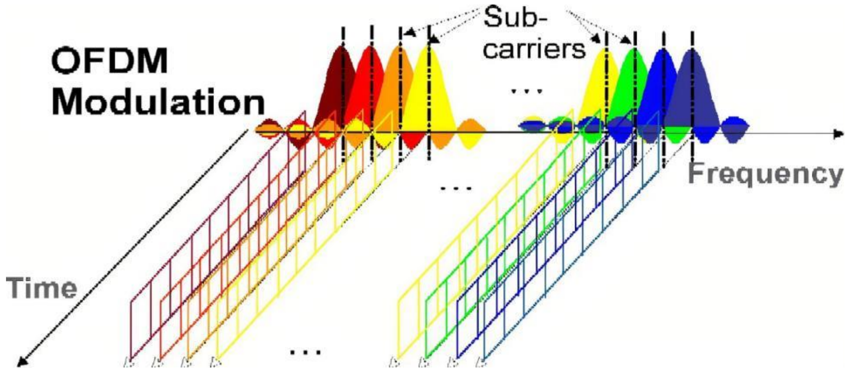
$$\Delta f = B/N$$

Let the first carrier frequency be f0, then the nth carrier frequency is

$$f_n = f_0 + (n-1)\Delta f$$

Modulate the symbol X_n to the nth carrier to get a transmitted symbol $X_n e^{j2\pi f_n t}$

$$f(t) = \sum_{n=1}^{N} X_n e^{j2\pi f_n t} = e^{j2\pi f_0 t} \sum_{n=1}^{N} X_n e^{j2\pi n \Delta f t}$$



Demodulation using orthogonality

$$\Delta f \int_0^{1/\Delta f} f(t) e^{-j2\pi f_n t} dt = X_n + \sum_{k \neq n} \Delta f \int_0^{1/\Delta f} X_k e^{j2\pi f_k t} e^{-j2\pi f_n t} dt = X_n$$

After sampling, the number L sample is:

$$\hat{f}(lT) = \sum_{n=1}^{N} X_n e^{j2\pi n \Delta f l T} = \sum_{n=1}^{N} X_n e^{j2\pi n \frac{B}{N} l \frac{1}{B}} = \sum_{n=1}^{N} X_n e^{j2\pi \frac{nl}{N}}$$

The equivalent of doing an IFFT

The block diagram of OFDM is as following:

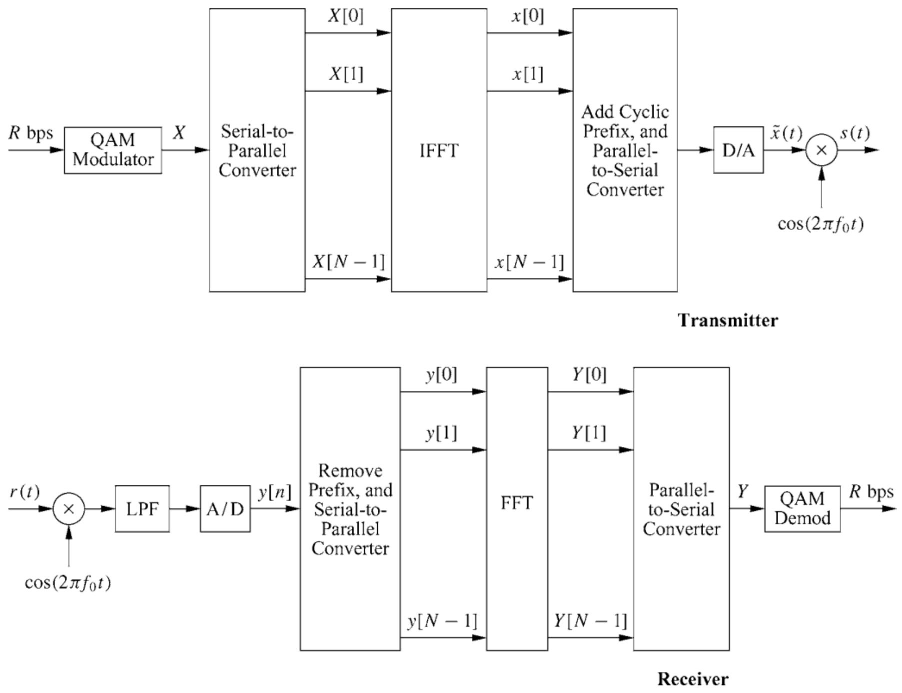


Fig1 the block diagram of OFDM

The OFDM implementation of multicarrier modulation is shown in Figure 12.7. The input data stream is modulated by a QAM modulator, resulting in a complex symbol stream $X[0], X[1], \ldots, X[N-1]$. This symbol stream is passed through a serial-to-parallel converter, whose output is a set of
N parallel QAM symbols $X[0], \ldots, X[N-1]$ corresponding to the symbols transmitted over each of the subcarriers. Thus, the N symbols output from the serial-to-parallel converter are the discrete frequency components of the OFDM modulator output $s(t)$. In order to generate $s(t)$, the frequency components are converted into time samples by performing an inverse DFT on these N symbols, which is efficiently implemented using the IFFT algorithm. The IFFT yields the OFDM symbol consisting of the sequence $x[n] = x[0], \ldots, x[N-1]$ of length N.

At the transmitter, using the QAM modulation method to modulate the digital message signal. Before transmitting it to a wireless channel, using the technique inverse discrete frequency transform (DFT) transform the representation of the inverse DFT. Then pass the signal to an equivalent wireless channel. Also, at the receiver, by using the DFT operation, inverse the IDFT operation at the transmitter and then do the QAM demodulation to recover the original signals.

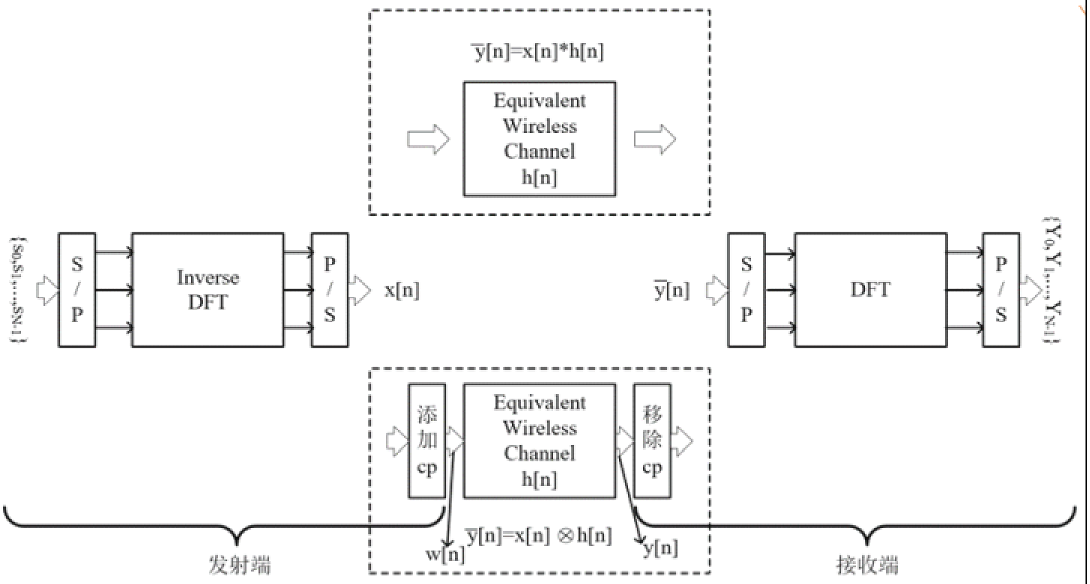


Fig2 the block diagram of OFDM with details

MATRIX Representation of OFDM

$$\begin{bmatrix} y_{N-1} \\ y_{N-2} \\ \vdots \\ y_0 \end{bmatrix} = \begin{bmatrix} h_0 & h_1 & \cdots & h_\mu & 0 & \cdots & 0 \\ 0 & h_0 & \cdots & h_{\mu-1} & h_\mu & \cdots & 0 \\ \vdots & \vdots & \ddots & \ddots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & h_0 & \cdots & h_{\mu-1} & h_\mu \end{bmatrix} \begin{bmatrix} x_{N-1} \\ \vdots \\ x_0 \\ x_{-1} \\ \vdots \\ x_{-\mu} \end{bmatrix} + \begin{bmatrix} \nu_{N-1} \\ \nu_{N-2} \\ \vdots \\ \nu_0 \end{bmatrix}$$

$$\begin{bmatrix} y_{N-1} \\ y_{N-2} \\ \vdots \\ \vdots \\ \vdots \\ y_0 \end{bmatrix} = \begin{bmatrix} h_0 & h_1 & \cdots & h_\mu & 0 & \cdots & 0 \\ 0 & h_0 & \cdots & h_{\mu-1} & h_\mu & \cdots & 0 \\ \vdots & \vdots & \ddots & \ddots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & h_0 & \cdots & h_{\mu-1} & h_\mu \\ \vdots & \vdots & \ddots & \ddots & \ddots & \ddots & \vdots \\ h_2 & h_3 & \cdots & h_{\mu-2} & \cdots & h_0 & h_1 \\ h_1 & h_2 & \cdots & h_{\mu-1} & \cdots & 0 & h_0 \end{bmatrix} \begin{bmatrix} x_{N-1} \\ x_{N-2} \\ \vdots \\ \vdots \\ x_0 \end{bmatrix} + \begin{bmatrix} \nu_{N-1} \\ \nu_{N-2} \\ \vdots \\ \vdots \\ \vdots \\ \nu_0 \end{bmatrix}$$

An alternate analysis for OFDM is based on a matrix representation of the system. H[n] is the frequency response of a low pass channel. V[n] is the noise caused while transmission. With the notation we can write the OFDM in the above matrix form.

4. IFFT/FFT:

Let x[n], 0 ≤ n ≤ N − 1, denote a discrete time sequence. The N-point DFT of x[n] is defined as:

$$DFT\{\mathrm{x}[n]\} = X[i] \triangleq \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} x[n] e^{-j2\pi ni/N}, 0 \le i \le N-1$$

The DFT is the discrete-time equivalent to the continuous-time Fourier transform, because X[i] characterizes the frequency content of the time samples x[n] associated with the original signal x(t). Both the continuous-time Fourier transform and the DFT are based on the fact that complex exponentials are eigen functions for any linear system. The sequence x[n] can be recovered from its DFT using the IDFT:

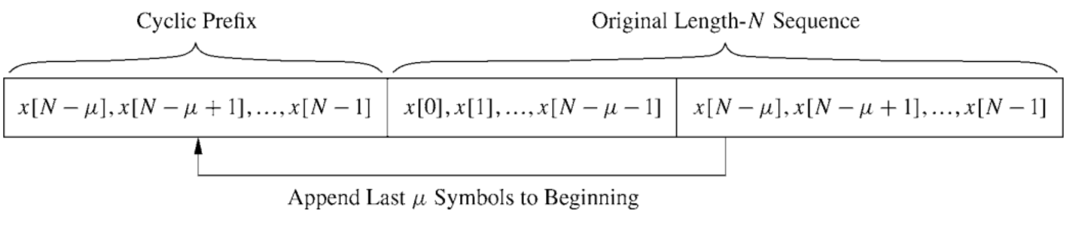
$$IDFT\{X[i]\} = x[n] \triangleq \frac{1}{\sqrt{N}} \sum_{i=0}^{N-1} X[i] e^{j2\pi ni/N}, 0 \le n \le N-1$$

The DFT and its inverse are typically performed via hardware using the fast Fourier transform (FFT) and inverse FFT (IFFT). When an input data stream x[n] is sent through a linear time-invariant discrete-time channel h[n], the output y[n] is the discrete-time convolution of the input and the channel impulse response:

$$y[n] = h[n] * x[n] = x[n] * h[n] = \sum_k h[k] x[n-k]$$

The N-point circular convolution of x[n] and h[n] is defined as

$$y[n] = x[n] \otimes h[n] = h[n] \otimes x[n] = \sum_k h[k] x[n-k]_N$$



where [n−k]N denotes [n−k] modulo N. In other words, x[n−k]N is a periodic version of x[n−k] with period N. It is easily verified that y[n] given by (12.16) is also periodic with period N. From the definition of the DFT, circular convolution in time leads to multiplication in frequency:

$$\mathrm{DFT}\{y[n] = x[n] \circledast h[n]\} = X[i] H[i], \quad 0 \le i \le N-1,$$

where $H[i]$ is the N point of $\{h[n]\}$. Note that if the sequence $\{h[n]\}$ is of length $k < N$

then it is padded with $N - k$ zeros to obtain a length N for the N-point DFT. If the channel and input are circularly convoluted then, as long as $h[n]$ is known at the receiver, the original data sequence $x[n]$ can be recovered by taking the IDFT of $Y[i]/H[i], 0 \le i \le N-1$. Unfortunately, the channel output is not a circular convolution but a linear convolution. However, the linear convolution between the channel input and impulse response can be turned into a circular convolution by adding a special prefix to the input called a cyclic prefix, described in the next section.

5. OFDM modulation & adding cyclic prefix

At the transmitter in OFDM system, the modulation makes the constellation of OFDM symbols in frequency domain.

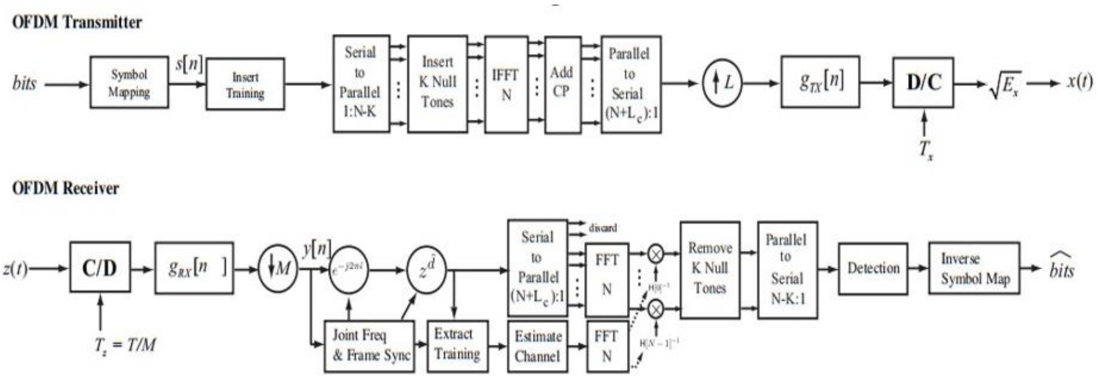


Fig3 the flow chart of OFDM at the transmitter & receiver

Assuming that K = 0, and the one OFDM symbol is consisted of N (2^p) frequency-domain symbols. And s[n] which n from n to N-1 is the cyclic prefix of length L_c.

The sequence after the modulation at transmitter is:

$$x[n] = IDFT\{s[n]\}$$

Notice that the s[n] adds the cyclic prefix of length L_c . This is because that going through a channel is consider as a linear convolution operation, that is:

$$y[n] = x[n] * h[n]$$

Where the x[n] is the transmitting signal, and the h[n] is the impulse response of the wireless channel.

At the receiver, we want the recovered signal can recover fully of the s[n], that is after the DFT operation:

$$DFT\{y[n],N\} = DFT\{x[n],N\}DFT\{h[n],N\}$$

Which can be simplified to:

$$DFT\{y[n],N\} = s[n]\ \ DFT\{h[n],N\}$$

However, the condition for these equations to be true is the s[n] is adding cyclic prefix of length L_c. This is because the multiplication of the DFT in frequency domain maps to the cyclic convolution in time domain. However, the cascade of a system is the linear convolution operation. So, it needs cyclic prefix to make the linear convolution to a cyclic convolution.

Then the transmitting signal can be expressed as:

$$w[n] = \frac{1}{N}\sum_{m=0}^{N-1} s[m]e\text{^}(j2\pi\, m(n - L\_c\ )/N)$$

Where the n is from 0 to $N + L\_c - 1$.

Also, notice that the $w[n] = w[n+N]$. Which is adding the cyclic prefix L_c to the original w[n]. It will ensure that there is no inter-symbol interference between adjacent OFDM symbols.

In conclusion, the CP provides two main functions, one of them is to against ISI to guards a better performance, while another is to convert the linear convolution to a circular convolution which leads to a more convenient demodulation.

6. Subcarrier and insert null tone

Multicarrier Modulation (MCM) employs multiple carrier signals. It decomposes the data stream into several sub-streams, so that the sub-streams have much lower transmission bit rates, and uses these data to modulate several carriers separately. Therefore, in a multi-carrier modulated channel, the data transmission rate is relatively low and the code element period is longer, which does not cause inter-code interference as long as the time delay extension is less than a certain ratio compared to the code element period. Thus, multi-carrier modulation is not sensitive to the time dispersion of the channel. The transmitter converts the transmitted digital signal into a mapping of subcarrier amplitude and phase and performs a discrete Fourier transform (IDFT) to change the spectral expression of the data to the time domain. The receiver performs the opposite operation to the transmitter, decomposing with the FFT transform, and the amplitude and phase of the subcarrier are finally converted back to a digital signal. An OFDM symbol includes within it a composite signal of multiple modulated subcarriers, each of which can receive modulation by PSK (phase shift keying)/QAM (quadrature amplitude modulation).

The OFDM transmitter maps the information bit stream into a PSK or QAM symbol sequence, and later converts the serial symbol sequence into a parallel symbol stream. Each N symbols that undergo serial-parallel conversion are modulated by a different subcarrier. So the number of our subcarriers depends on the number of N of IFFT/FFT.

An OFDM symbol is a composite signal of N parallel symbols, and if the transmission time (period) of a single serial symbol is Ts, the duration (period) of an OFDM symbol T_{sym} = N*Ts. The frequency domain modulated signal X[k] has the frequency: f_k = k/T_{sym} and the number of subcarriers is N. Then k = 0, 1, 2, ..., N-1.

The subcarrier number affected the transmission rate, because a single sub-carrier OFDM transmission rate equals the transmission rate of single subcarrier multiplies general subcarrier number.

$$TransmissionRate = transmissionRate/per\ subcarrier * subcarrier\ number$$

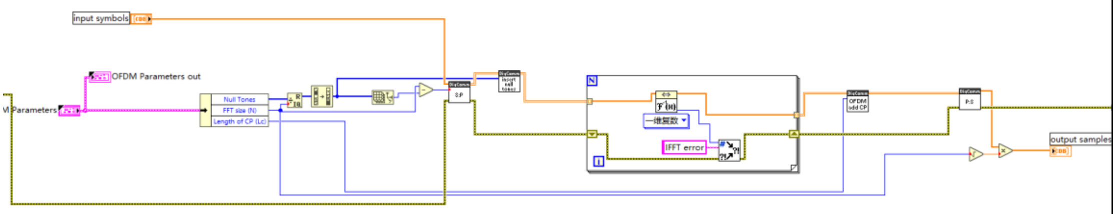
However, as the number of subcarriers increases, the sensitivity of OFDM to frequency bias increases for a limited spectrum. This is because as N increases, the spectrum occupied by a single carrier decrease, the frequency spacing between carriers is smaller, and for the same frequency offset, the resulting inter-subcarrier interference (ICI) is greater, so the sensitivity to frequency bias increases.

Normally, the size of IFFT should equals to the N power of 2, because of the characteristic of FFT. However, in the real transmission, the number of the subcarriers can vary. As a result, we need to add null tone. The main purpose of the null carrier insertion of the null tone is to satisfy the requirement to perform the discrete Fourier transform N when the transmitted signal is less than the number of subcarriers of 2^m(i.e., it must be a number to the power of 2).

Lab results & Analysis:

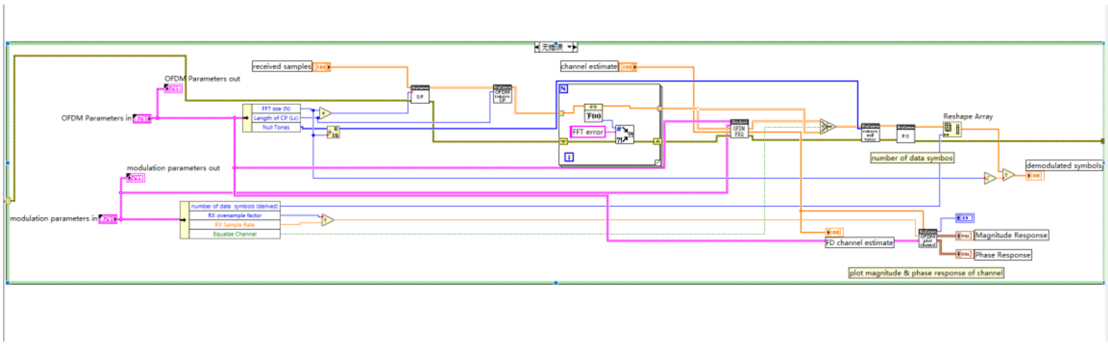
OFDM Block Diagram

OFDM Modulator:



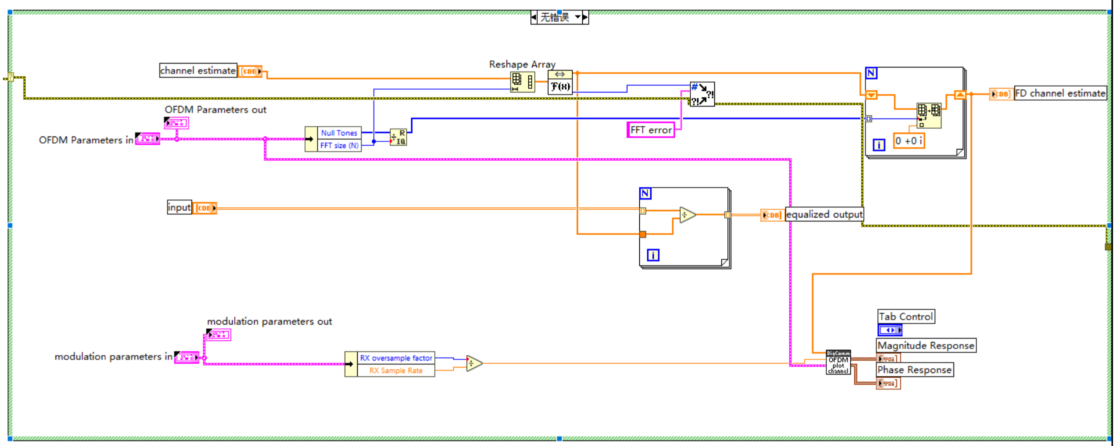
In the modulation, first we transform the serial signal into parallel for the IFFT. When the IFFT allows N inputs, the number of allowed input symbols is $M = N - n$, where n is the number of null tones. Then, if the number of input symbols is not divisible by M, we pad zeros at the end of the symbol sequence. Then we convert the symbol sequence into several parallel signal with symbols by the function Reshape Array, and do the subsequent operation separately. Then we insert null tones according to the given indexes of null tones using the function Insert into Array, and do the IFFT separately to each parallel signal. After that, get the last several symbols of each parallel signal and add them to the front as the CP, and reshape it again to serial signal. At last, we need to do a normalization cause the IFFT will change the total power of the signal.

OFDM Demodulator:



In the demodulation, we transform the signal into parallel form as we do in the modulation. Then we delete first several symbols of each parallel signal as the CP, and do the FFT to get the signals in frequency domain. Then we need to delete the null tones. Because the indexes of later symbols will change if the former ones are deleted, we should sort the given null tone indexes decreasingly and do the null tone deletion. Next, transform the signal into serial and remove the padding zeros according to the length of the data symbols. Similarly, a normalization is needed at last due to the effect of signal power of FFT.

OFDM FEQ:



In the previous lab, we have already learned how to get the time domain estimation of the channel. However, in OFDM, we want the channel information in frequency domain, that is because we use IFFT to modulate the signal, the information of the signal is stored in the frequency domain. So what we should do is just to FFT the estimated channel information and then divide by the original signal. The process of frequency domain equalization involves taking the results obtained from channel estimation, performing FFT to transform them into the frequency domain, and then dividing the input signal by this value. This allows us to obtain the original input signal, completing a simple frequency domain equalization algorithm.

OFDM Technology Frequency Bias Sensitivity

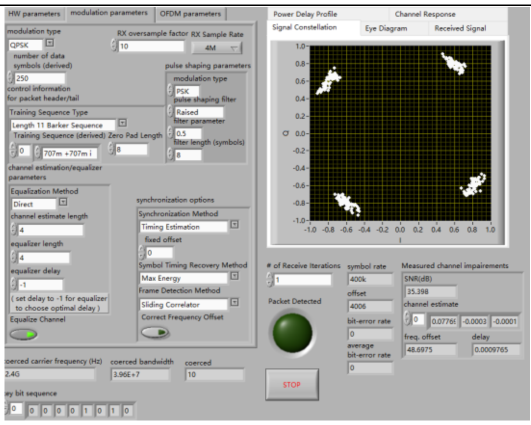
Use usrp to find the Frequency Bias Sensitivity

$$Phase\ deg. = 2 \cdot 180^{\circ} \cdot f_o \cdot n_{max} \cdot \frac{1}{f_s}, \qquad n_{max} = 299, f_s = 1M/s$$

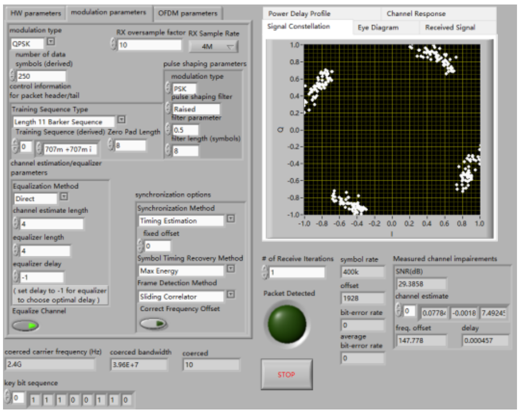
The phase degree is dependent of frequency offset and the sample rate.

We set that Oversample Factor=10,f_s=4M/s , and adjust N(fft size) and cp

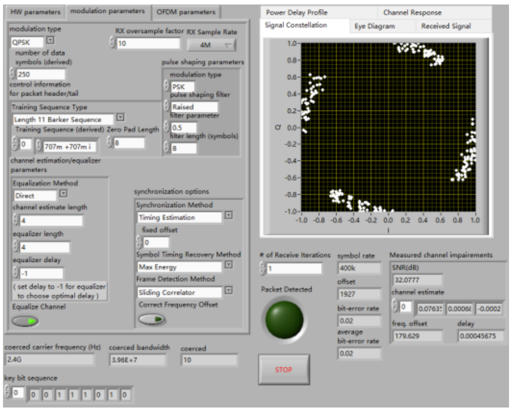
When N=64,cp=8



50Hz

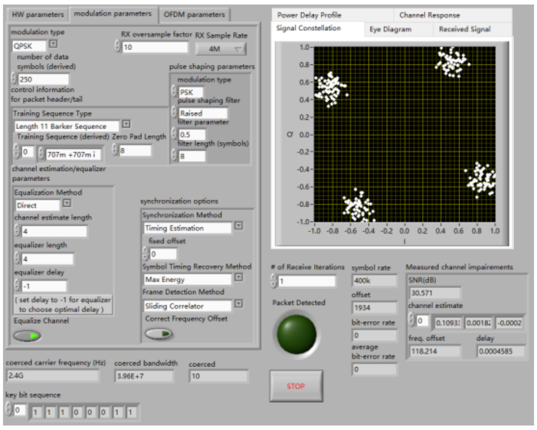


100Hz

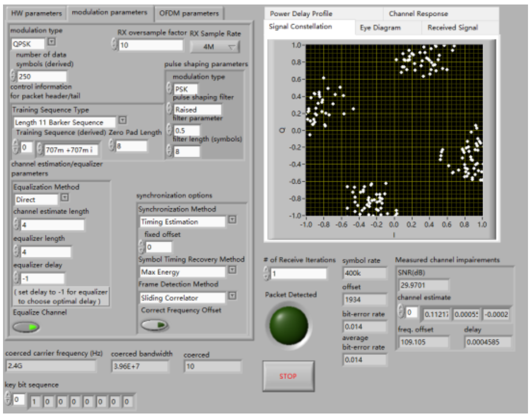


150Hz

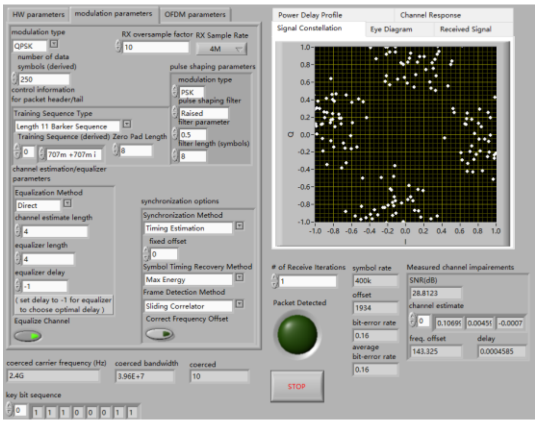
N=512 cp=16



50Hz

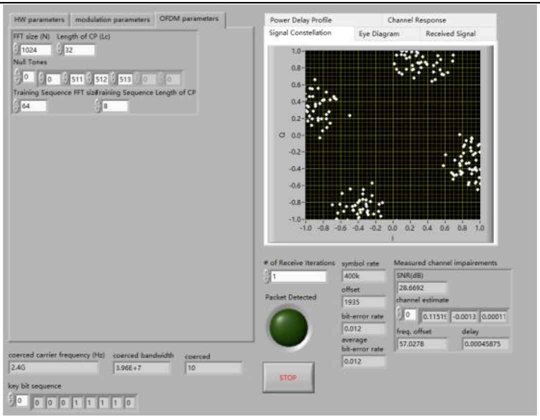


100Hz

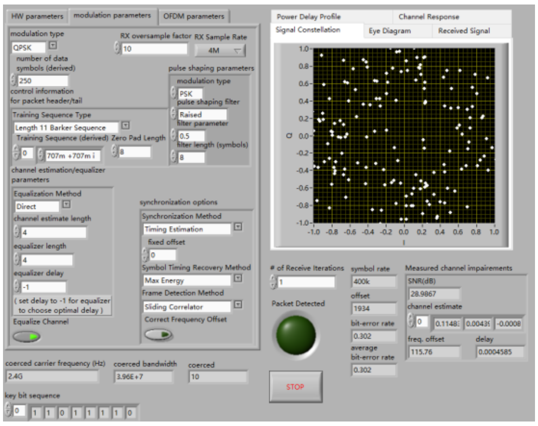


150Hz

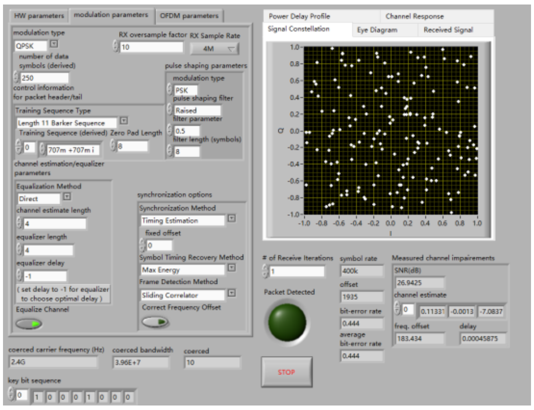
N=1024 cp=32



50Hz



100Hz



150Hz

From the figures above, it's concluded that as the frequency offset increased, the constellation points disperse, which means that the equalizer does not works well. In the other hand, for different N parallel symbols at the same frequency offset situation, the constellation manifests worse at a large N.

The subcarrier orthogonality of OFDM system can be destroyed by distortion and doppler shift frequency offset. Its effect is also called ICI, and its effect is huge as only 1% of the frequency offset will cause the loss of SNR 30 db.

Mathematically analysis of the sensitivity of the frequency offset in OFDM systems.

Assuming the subcarrier number of OFDM system is N, the original input signal is $\{a_k\}$.

Then the transmitting signal can be:

$$x(t) = e^{j2\pi f_c t} \sum_{k=0}^{N-1} IDFT\{a_k\} \phi(t - kT/N)$$

Where the $\phi\left(t - \frac{kT}{N}\right)$ satisfies the condition of Nyquist rules.

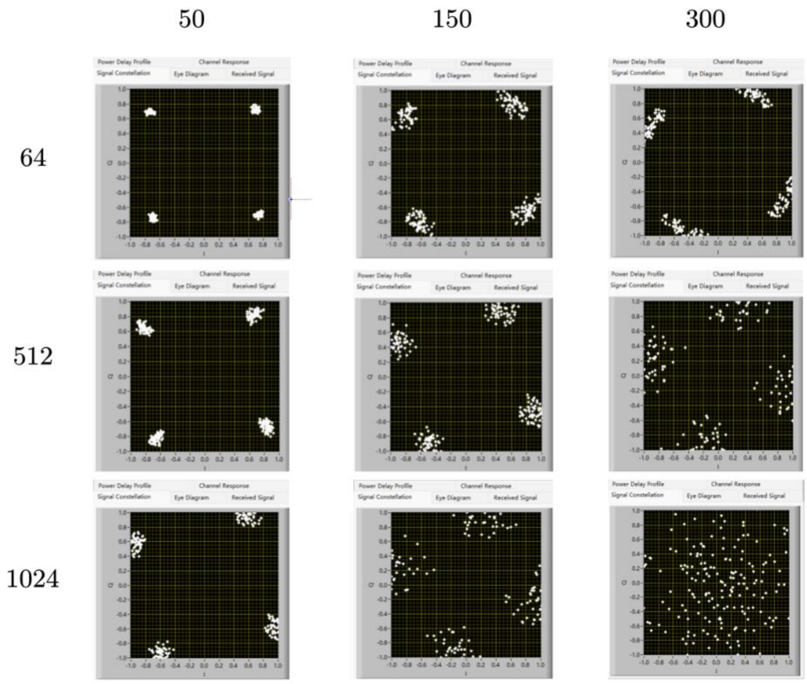
When the receiver has the frequency offset Δf. The received signals can be expressed as:

$$y(t) = e^{j2\pi \Delta f t + \theta_0} \sum_{k=0}^{N-1} IDFT\{a_k\} p(t - kT/N)$$

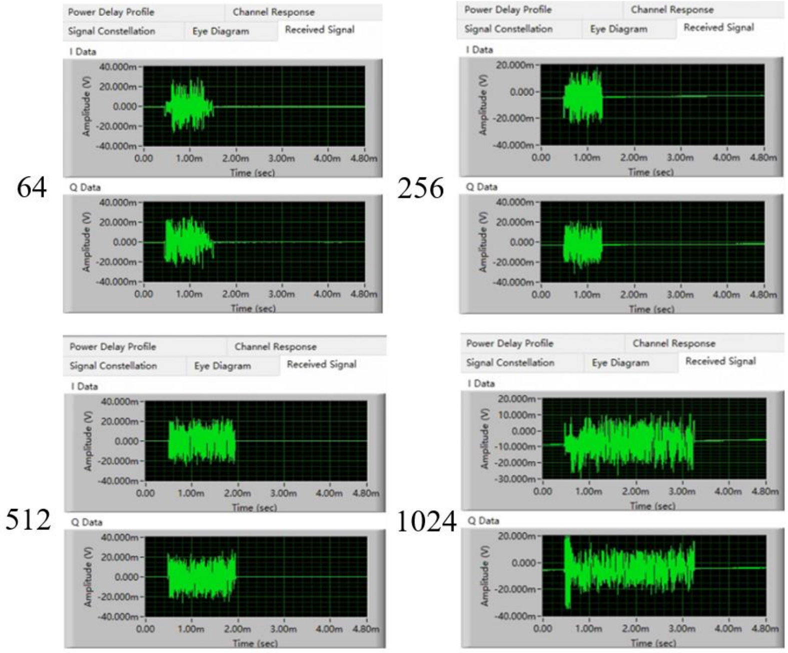
Where the $p\left(t - \frac{kT}{N}\right)$ is a function which related to Δf. when Δf is not equal to zero, the recovered $\{a_k\}$.is influenced by the Δf. That's why the frequency offset effect the SNR. And for large N, the effect is more evident. This experiment and insight also reveal that there is a tradeoff between N and better performance at the receiver. Which means that there is a balance between large parallel number or the faster transmission speed and better performance.

The Impact of the Subcarrier Numbers on the OFDM System

We will explore the impact of the subcarrier numbers on the OFDM system, based on the results explored in the previous section regarding the impact of OFDM on frequency offset, we can illustrate that the number of subcarriers has a significant influence on frequency offset sensitivity. Combining the previous results, the findings are presented in the following graph. FFT size represents the number of the subcarriers. From the graphs, we can see that when the FFT size is larger, the symbols are more separated and more sensitive to frequency offset and there is a higher possibility to lead to bit error. When the frequency offset is 300Hz, there is no error bits when FFT size is only 64. But the symbols when FFT size is 512 and 1024 are too separated to recover the correct bits. Increment of both FFT size and frequency offset will incur a sparser signal constellation. Therefore, although increasing the number of subcarriers can make the channel flat for each carrier and increase the transmission rate, in OFDM systems, we cannot indiscriminately increase the number of subcarriers, as it will make the system extremely sensitive to frequency offset.



On top of that, we can observe that with an increase in the number of subcarriers, the bandwidth of each subcarrier decreases. This implies that the time-domain duration of the signal to be transmitted becomes larger. From the experimental results shown in the figure below, it can be seen that as the number of subcarriers increases, the time-domain length of the received signal gradually increases.

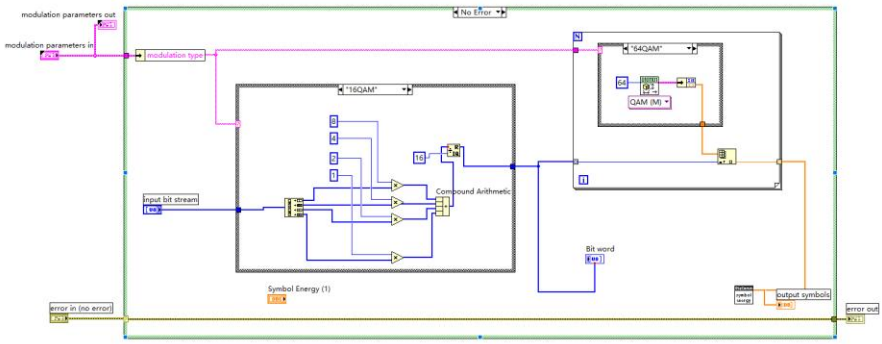


Advance:

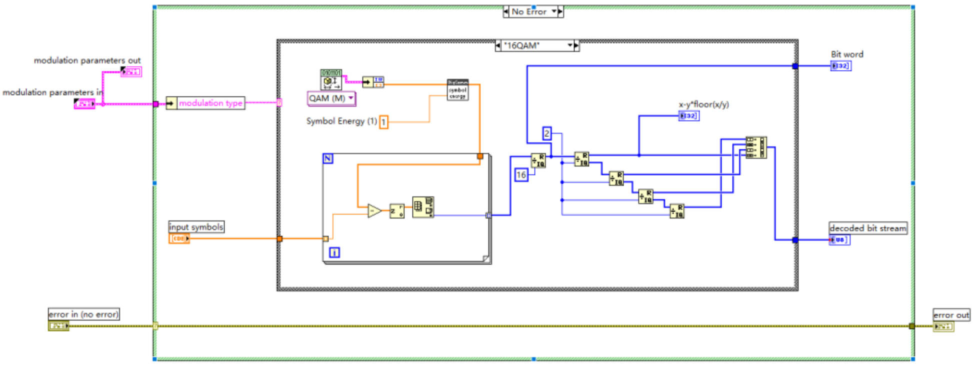
1.Higher order modulation of subcarriers

We need to modify the encoding module in front of OFDM modulation and the decoding module behind OFDM demodulation to change the bit stream to symbol stream. The same encoding and approach as the first experiment is done by converting the binary to decimal by the number of encoded bits and then going through the symbol mapping table to complete the encoding. The decoding is then done by going through the symbol mapping table and then converting the decimal number to a bit stream.

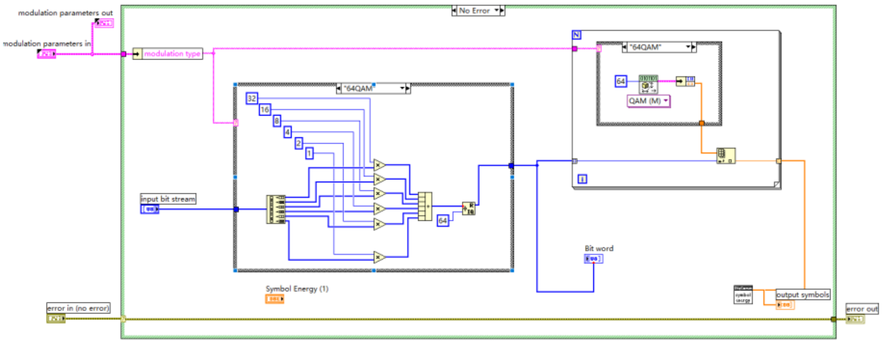
16QAM encode:



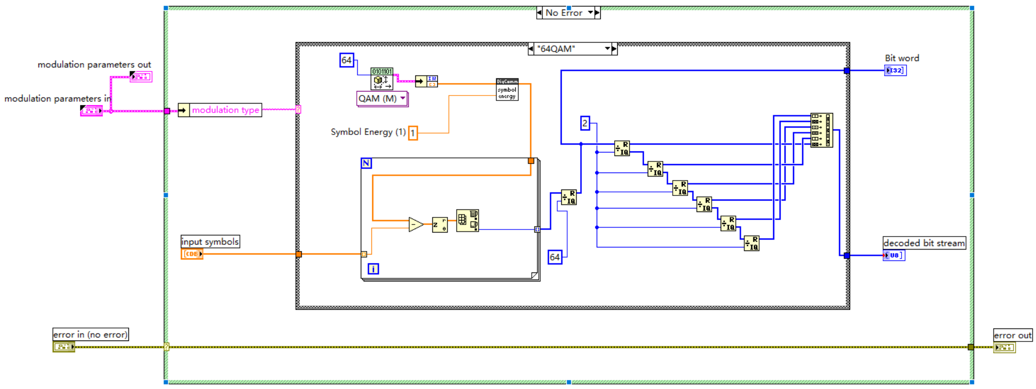
16QAM decode:



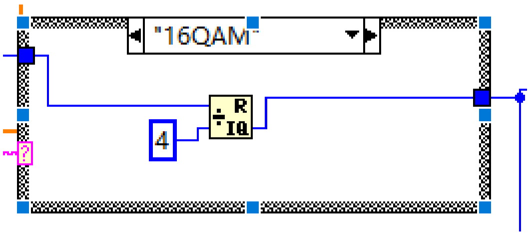
64QAM encode:

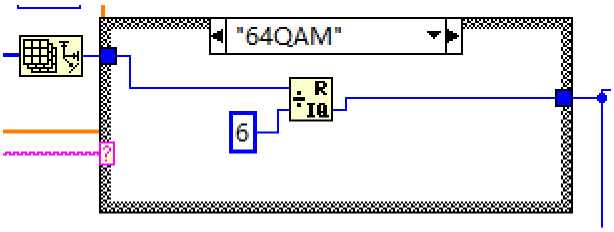


64QAM decode:



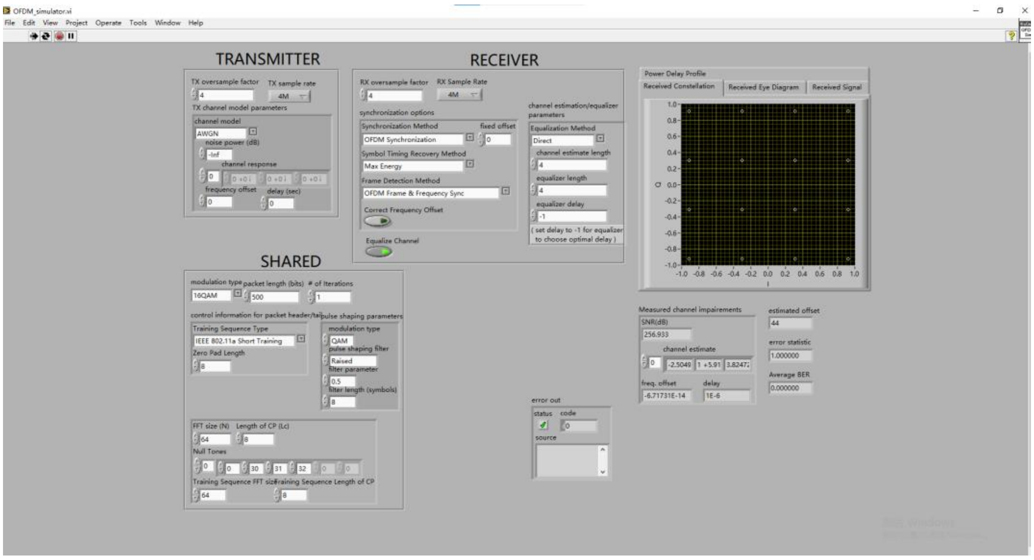
Also, we need to change the calculation within RX TX inti for the number of symbols. Otherwise, it will not be possible to delete the zeros that are made up during the string to soldier conversion. If it is not deleted, it will lead to constellation points at the origin and make the system decode wrongly.





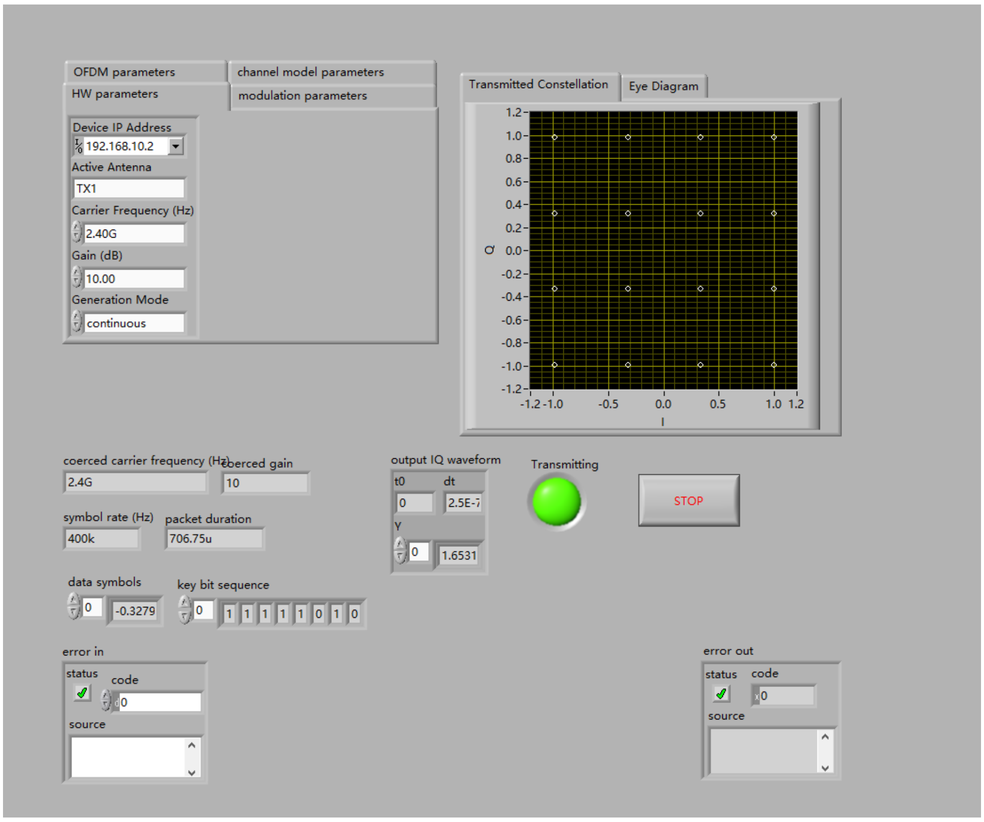
Result:

16QAM

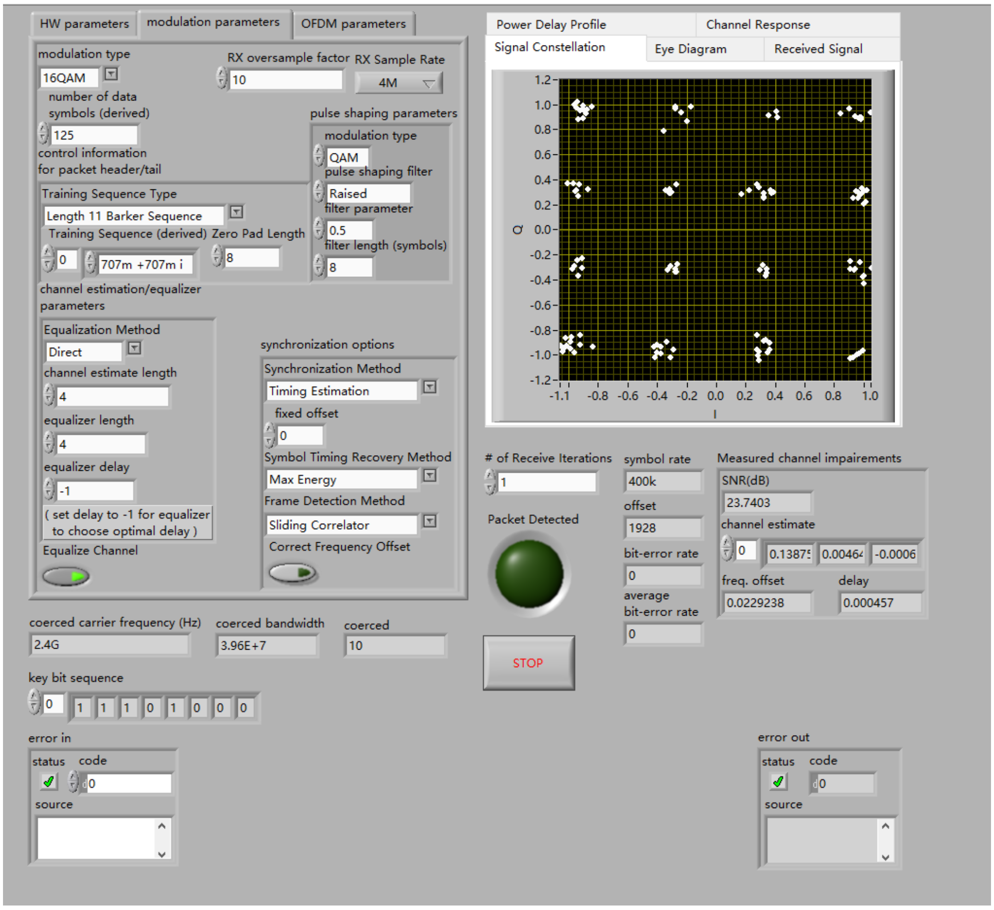


USRP

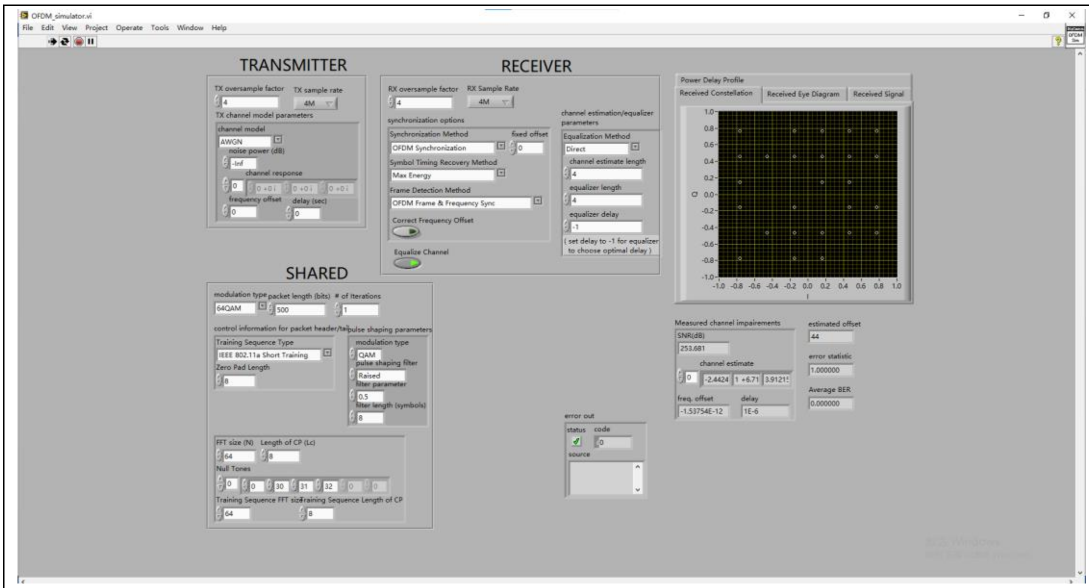
TX


OFDM parameters
channel model parameters
HW parameters
modulation parameters
Device IP Address
192.168.10.2
Active Antenna
TX1
Carrier Frequency (Hz)
2.40G
Gain (dB)
10.00
Generation Mode
continuous
Transmitted Constellation
Eye Diagram
Q
I
coerced carrier frequency (Hz)
2.4G
coerced gain
10
symbol rate (Hz)
400k
packet duration
706.75u
output IQ waveform
t0
dt
0
2.5E-7
Y
0
1.6531
Transmitting
STOP
data symbols
0
-0.3279
key bit sequence
0
1 1 1 1 1 0 1 0
error in
status
code
0
source
error out
status
code
0
source

RX

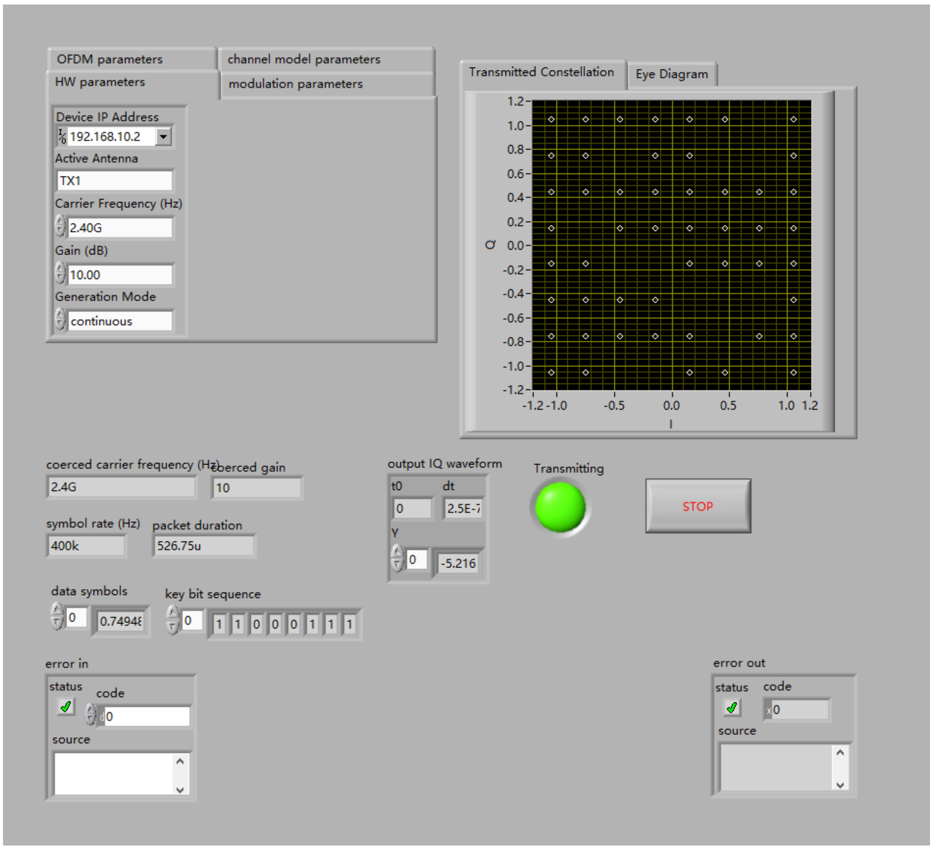

HW parameters
modulation parameters
OFDM parameters
modulation type
16QAM
RX oversample factor
10
RX Sample Rate
4M
number of data symbols (derived)
125
control information for packet header/tail
pulse shaping parameters
modulation type
QAM
pulse shaping filter
Raised
filter parameter
0.5
filter length (symbols)
8
Training Sequence Type
Length 11 Barker Sequence
Training Sequence (derived)
0
707m +707m i
Zero Pad Length
8
channel estimation/equalizer parameters
Equalization Method
Direct
channel estimate length
4
equalizer length
4
equalizer delay
-1
(set delay to -1 for equalizer to choose optimal delay)
Equalize Channel
synchronization options
Synchronization Method
Timing Estimation
fixed offset
0
Symbol Timing Recovery Method
Max Energy
Frame Detection Method
Sliding Correlator
Correct Frequency Offset
Power Delay Profile
Channel Response
Signal Constellation
Eye Diagram
Received Signal
Q
I
# of Receive Iterations
1
symbol rate
400k
offset
1928
bit-error rate
0
average bit-error rate
0
Measured channel impairements
SNR(dB)
23.7403
channel estimate
0
0.13875
0.00464
-0.0006
freq. offset
0.0229238
delay
0.000457
Packet Detected
STOP
coerced carrier frequency (Hz)
2.4G
coerced bandwidth
3.96E+7
coerced
10
key bit sequence
0
1 1 1 0 1 0 0 0
error in
status
code
0
source
error out
status
code
0
source

64QAM:

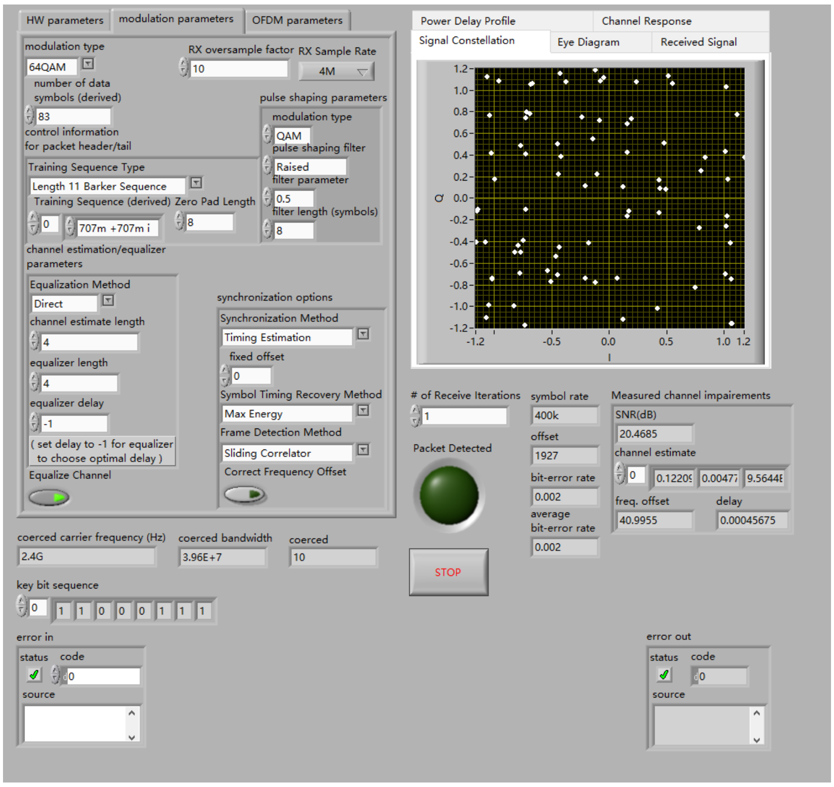

TRANSMITTER
RECEIVER
SHARED

USRP

TX


OFDM parameters
channel model parameters
HW parameters
modulation parameters
Device IP Address
192.168.10.2
Active Antenna
TX1
Carrier Frequency (Hz)
2.40G
Gain (dB)
10.00
Generation Mode
continuous
Transmitted Constellation
Eye Diagram
coerced carrier frequency (Hz)
2.4G
coerced gain
10
symbol rate (Hz)
400k
packet duration
526.75u
output IQ waveform
t0
dt
0
2.5E-7
Y
-5.216
Transmitting
STOP
data symbols
key bit sequence
error in
status
code
source
error out

RX

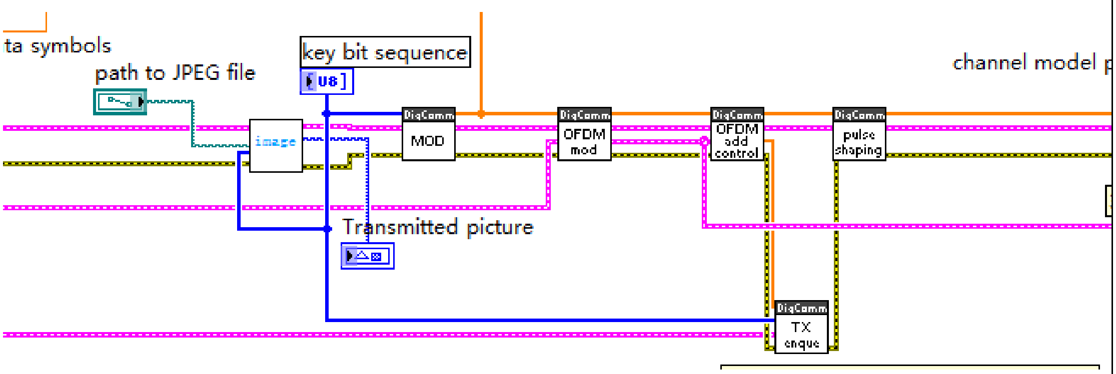


It can be seen that we have performed the verification of the higher order subcarrier modulation system using USRP.

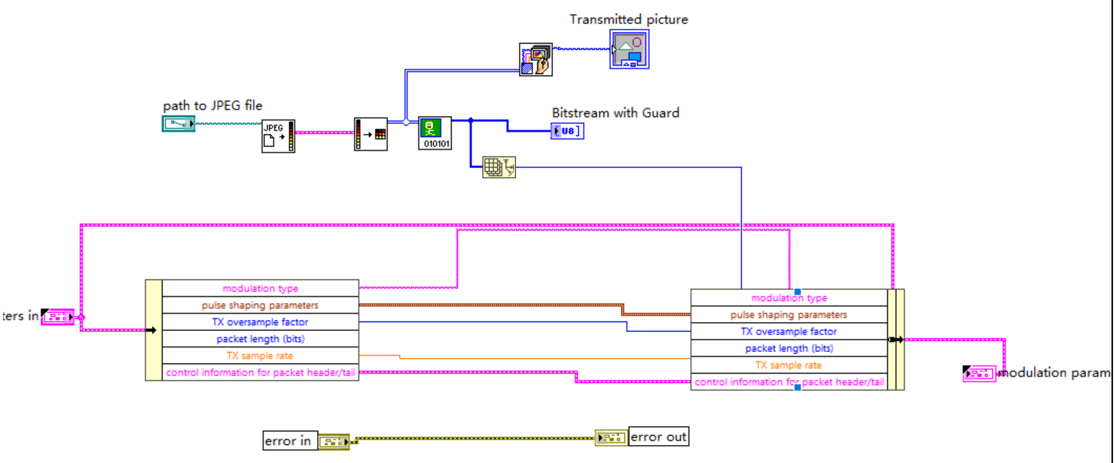
2. Image Transmission

TX:

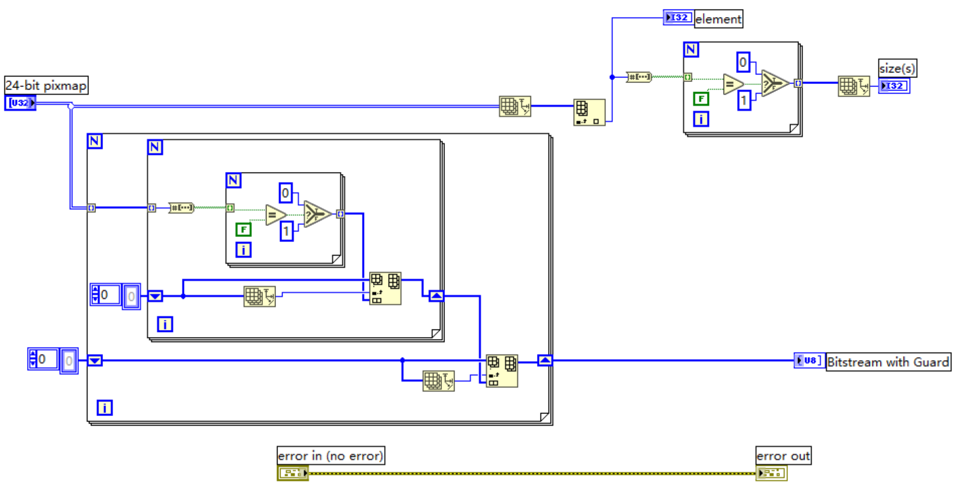
We learned that the source.vi is to generate random bits for transmission. So, we replace this vi with the program we built to input the bits from images. Our vi is image2bits.vi whose position in the entire program is shown as below:



This vi is used to convert the image to the corresponding bitstream to transmit further. The specific details are shown as below which use the JPEG module in Labview to read the image file and then use the 32bits2stream.vi to convert the 32bit generated to the bitstream.

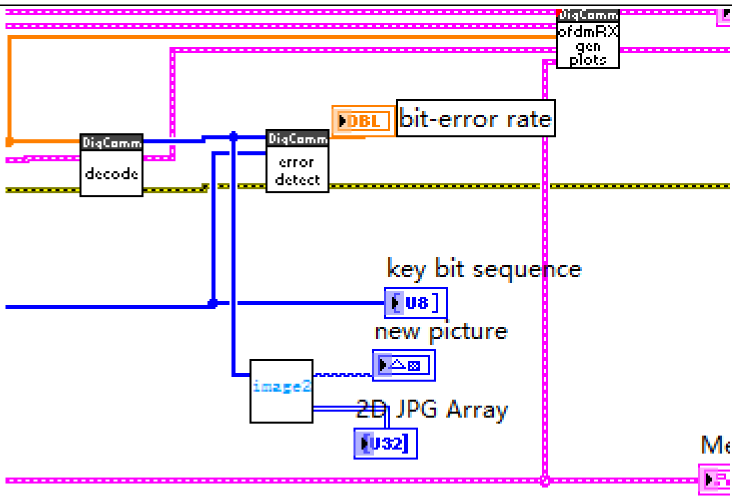


The specific details of 32bit_to_bitstreamV2.vi are shown as below:

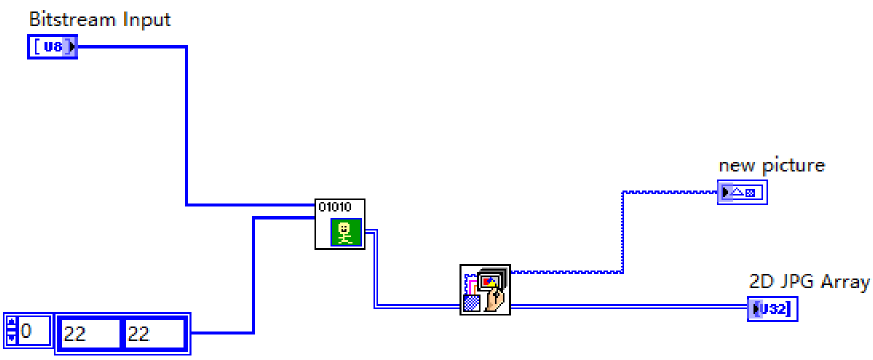


RX:

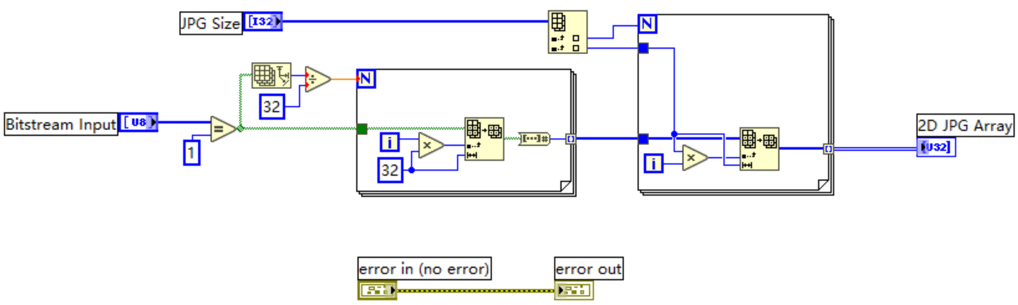
For the RX, the process is the reverse of the previous transmission process. It involves taking the bits demodulated by the decode module and converting them back into an image format for output. The conversion process is implemented through a sub VI that we have developed, and its position in the overall program is as follows:



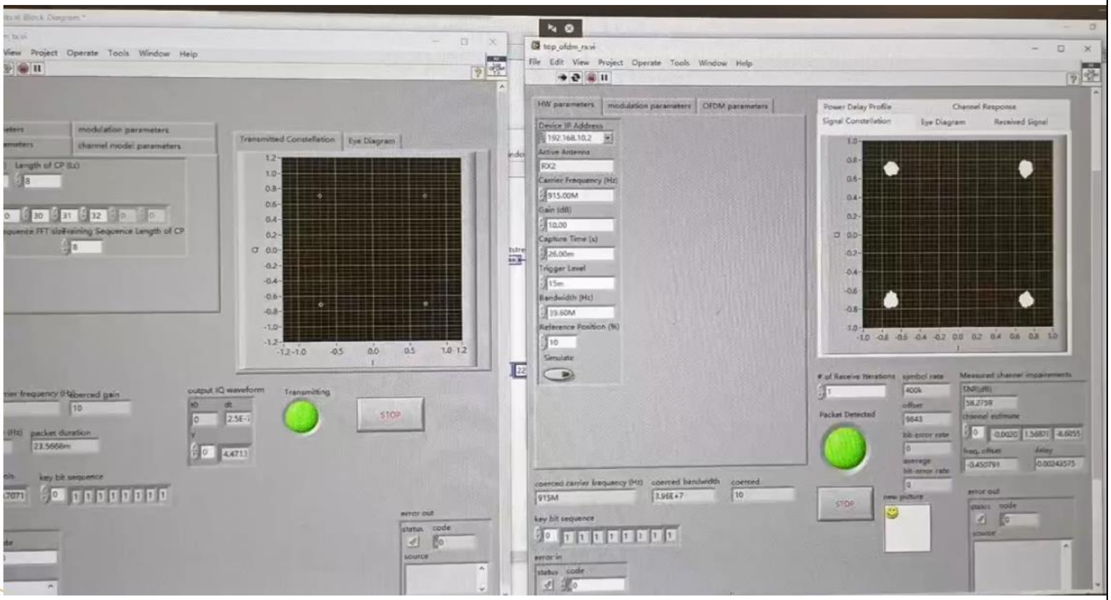
The details of the bits2image.vi are shown as below:



The process is simple which use the bitstream_to_32bitv2.vi to convert the input bitstream, that is the decoded bitstream to the recovered image. The details of the used bitstream_to_32bitv2.vi are shown as follows:



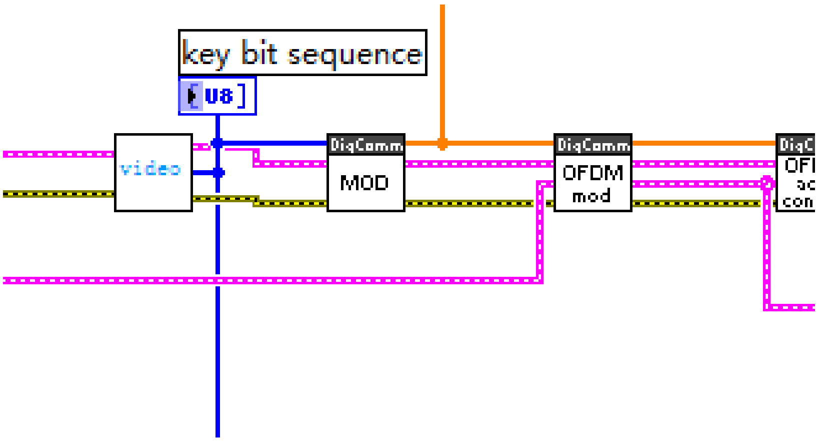
Then, as a result, the image can be recovered correctly and the bit error rate is close to zero which indicates that our program of the image-transmitting is correct.



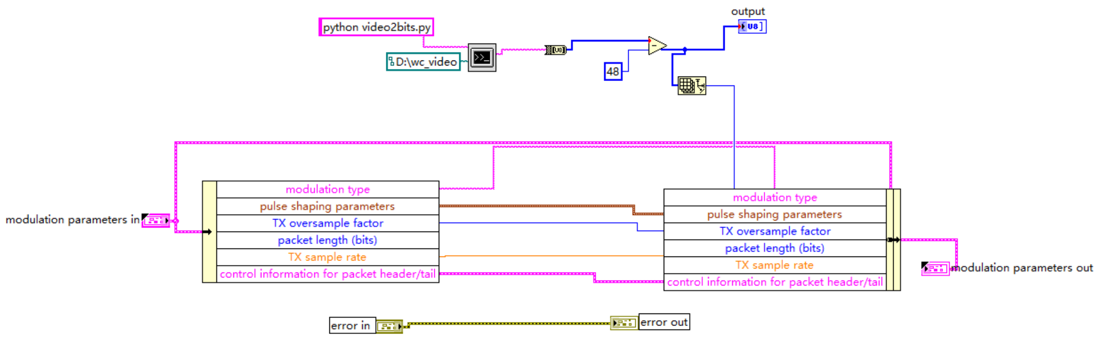
Video Transmission

TX:

Similarly, we can also perform video transmission. First, replace the source module with our video2bits module, which converts the selected video into a bitstream. The position of this module in the entire program is shown in the following figure:



The internal details of the video2bits are shown as below.

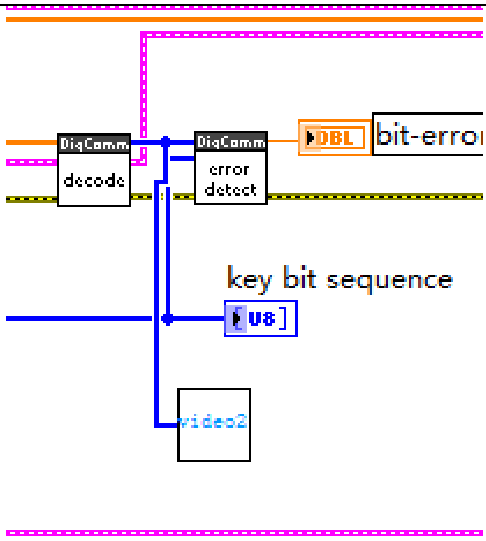


Because LabVIEW lacks relevant video processing modules, here we use system modules to call command-line execution of related Python files to convert the video into a bitstream. The relevant Python program is shown in the figure below. It's worth noting that the results obtained by this program are in ASCII code, so 0 corresponds to 48, and 1 corresponds to 49. Therefore, the converted result should be subtracted by 48 to obtain the correct binary bitstream.

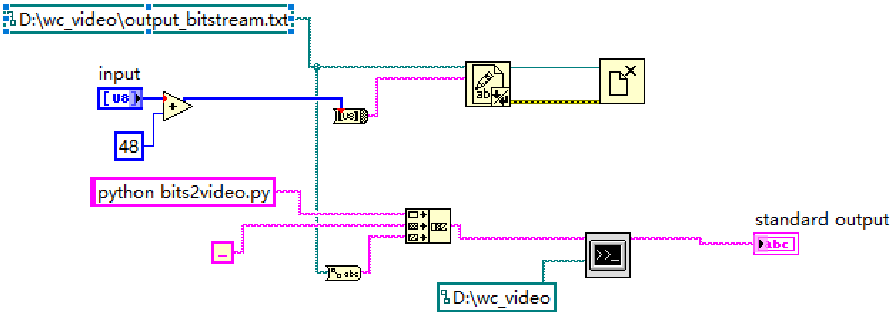
```
import struct
video_path = 'video.mp4'
output_path = 'output_bitstream.txt'
binary_data_list = []  # 用于存储每一帧的 binary_data
with open(video_path, 'rb') as video_file:
    # 逐帧读取视频
    frame_number = 0
    while True:
        frame_data = video_file.read(1)  # 逐字节读取
        if not frame_data:
            break
        # 将字节数据转换为二进制字符串
        binary_data = format(frame_data[0], '08b')
        # 存储每一帧的 binary_data
        binary_data_list.append(binary_data)
        frame_number += 1
    # 使用 join 将列表中的二进制数据连接成一个字符串
    binary_data_str = ''.join(binary_data_list)
    binary_data_str_cleaned = binary_data_str.replace(' ', '').replace(',', '').replace('[', '').replace(']', '')
    print(binary_data_str_cleaned)
```

RX:

For the RX, the process is the reverse of the previous transmission process. It involves taking the bits demodulated by the decode module and converting them back into an video format for output. The conversion process is implemented through a sub VI that we have developed, and its position in the overall program is as follows:



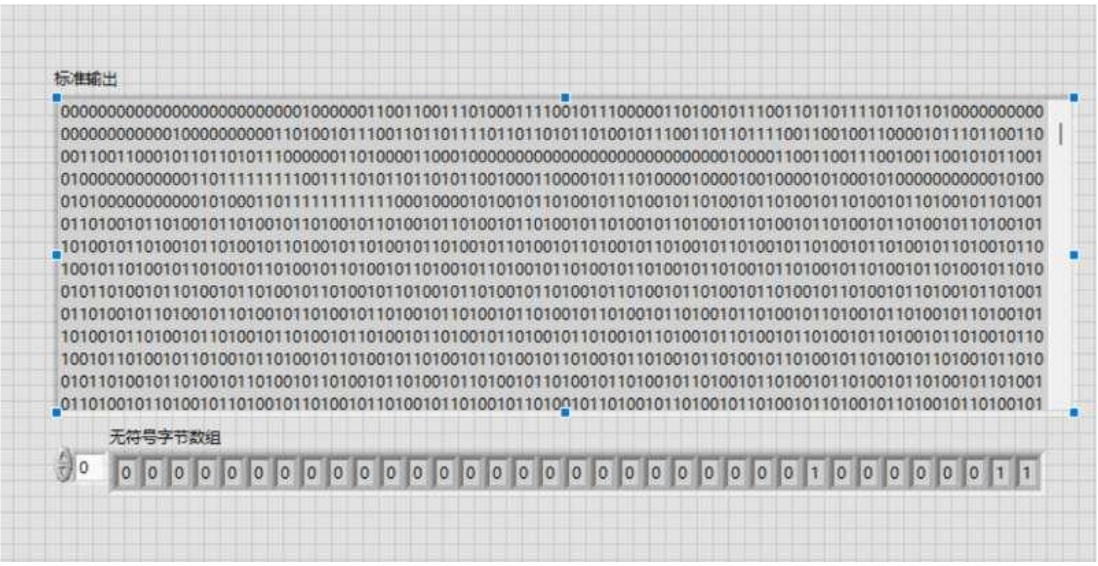
The internal details of the bits2video.vi are shown as below:



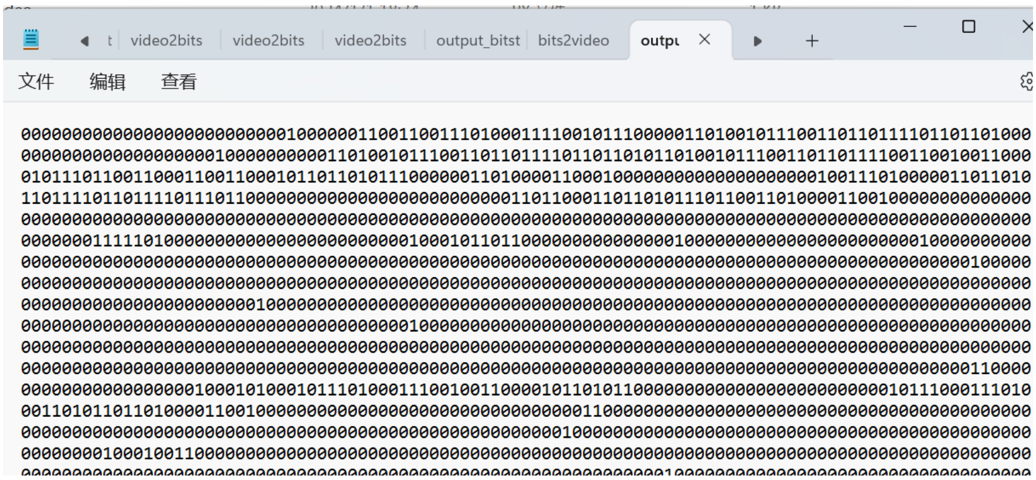
The idea is similar; we use system modules to call command-line execution of relevant Python files. First, the received binary bits (01) are stored in a text document. Then, the corresponding Python function is called to convert the binary bits from the file into the corresponding video. The relevant Python program is shown in the figure below. Similarly, before this step, it is necessary to convert the received binary bits plus 48 into ASCII code.

```python
import struct

input_path = 'output_bitstream.txt'
output_path = 'decoded_video.mp4'

with open(input_path, 'r') as input_file:
    binary_data = input_file.read()

# 将二进制字符串转换为字节对象
byte_data = bytes([int(binary_data[i:i+8], 2) for i in range(0, len(binary_data), 8)])

# 将字节数据写入视频文件
with open(output_path, 'wb') as output_file:
    output_file.write(byte_data)

print("Decoding complete.")
```

The standard output of the diagram are shown as below, it is recovered 01 bitstream of the video.



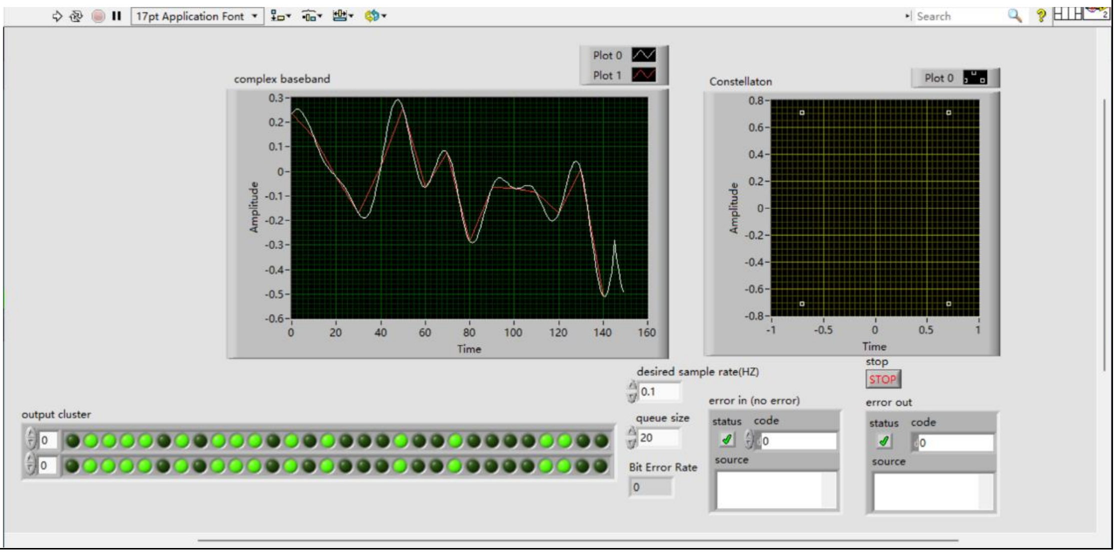
The output_bitstream.txt are:

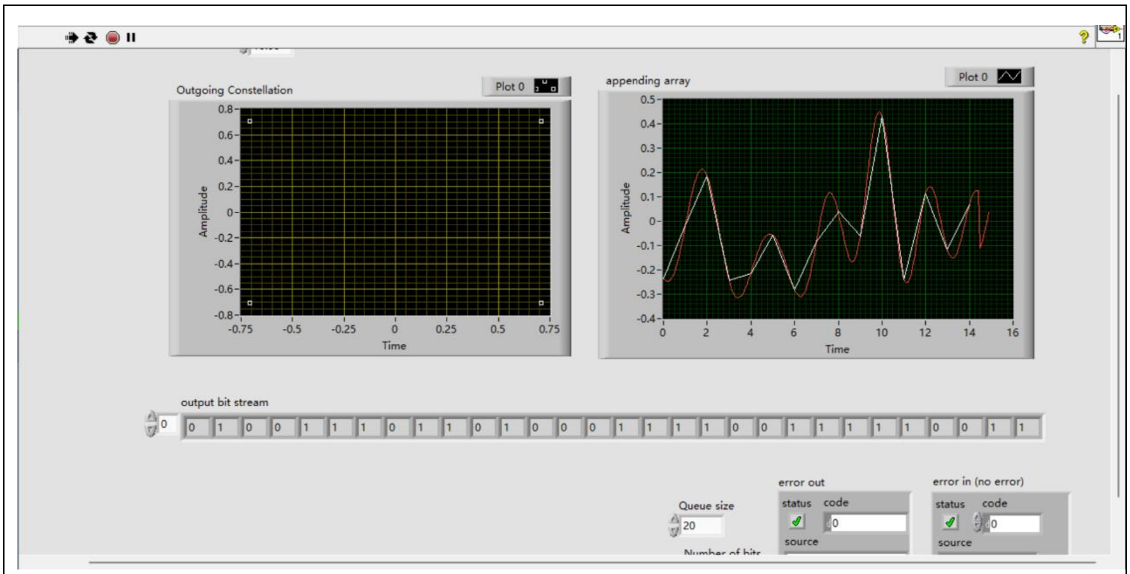


The recovered video is:

Experience

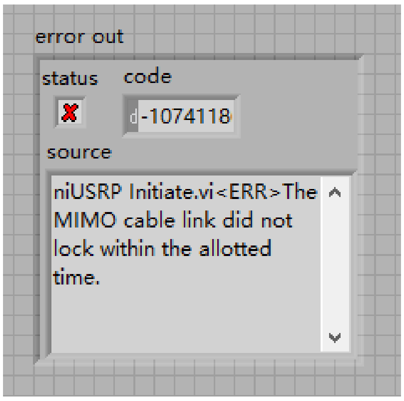
1. In-class lab screenshot



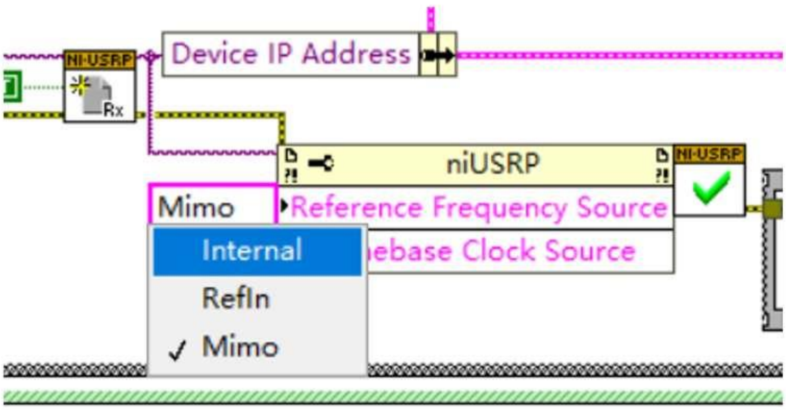


2. Problem we meet and experience

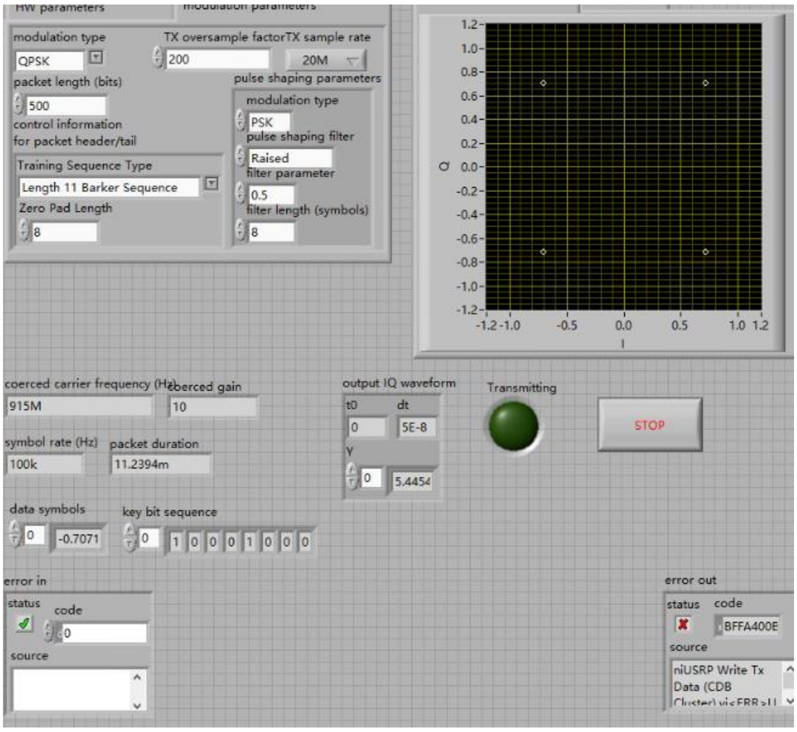
1. When we run the RX program, we will find that there is an error shown below which indicates that we use the MIMO cable clock instead of internal clock.



So, we enter into the initiate module to change MIMO to internal:

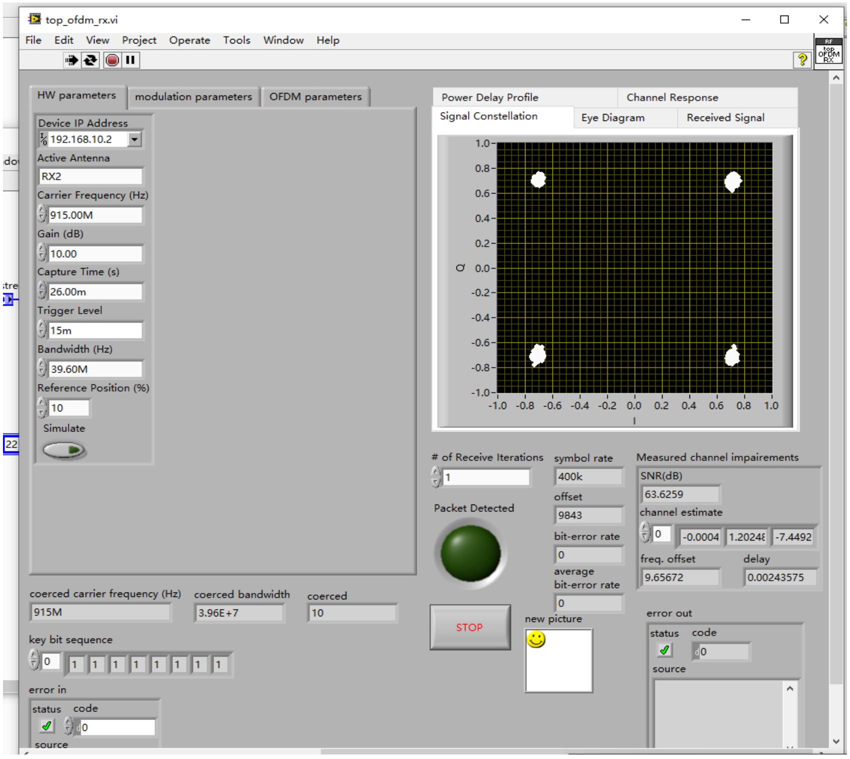


2. If the TX sample rate is too large, there is an error like this:

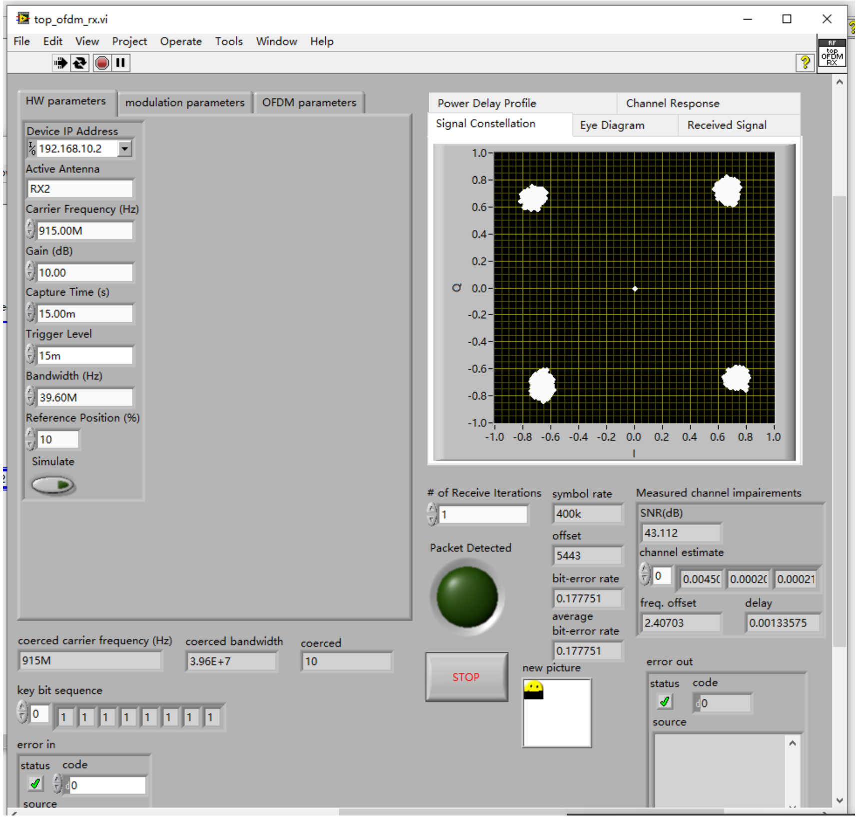


We can choose smaller TX sample rate. For example, we can use TX oversample factor of 10, the TX sample rate of 4M, which is proved to be errorless by practice.

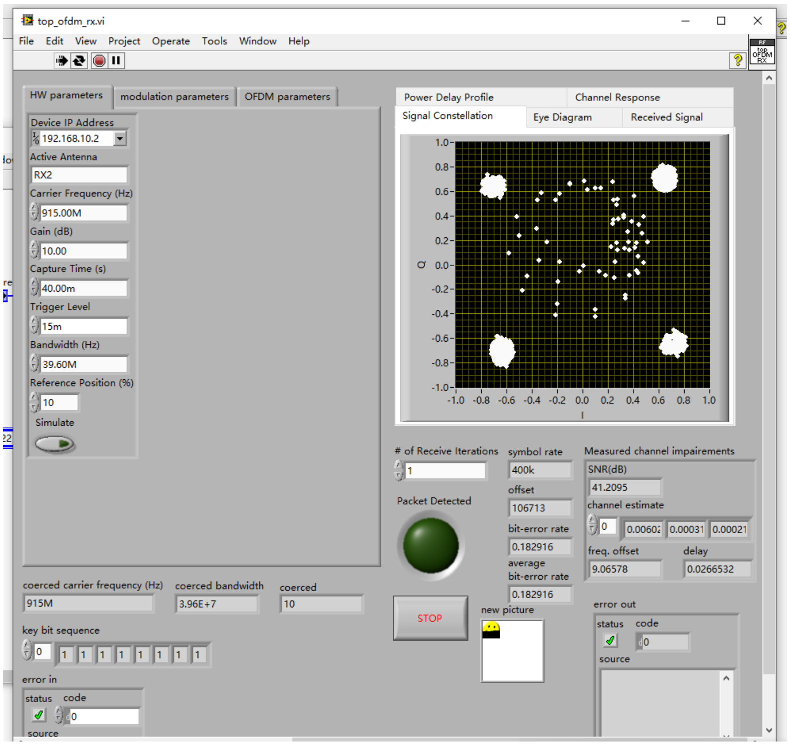
3. The capture time plays an important role in the image transmission, in result above, we choose the Capture time as 26ms, the result are well:



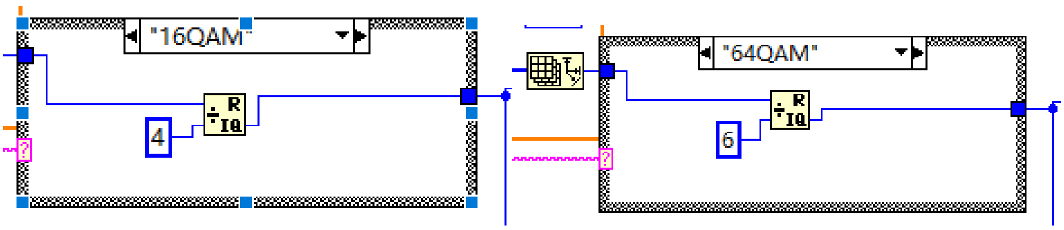
But when the capture time is smaller, it can not recover the entire image, because the capture time is not enough to capture all image bits. The result of Capture time as 15ms is shown as below:



However, the result is also not good when the capture time is too larger, because it will capture more noise as the capture time is increasing. The result of the Capture time as 40ms is shown as below:



4. we need to change the calculation within RX TX inti for the number of symbols. Otherwise, it will not be possible to delete the zeros that are made up during the string to soldier conversion. If it is not deleted, it will lead to constellation points at the origin and make the system decode wrongly.



3. Respective contributions

Song Yihang finish the introduction including the basic principle and corresponding knowledge of OFDM and explore the OFDM technology frequency bias sensitivity and complete the advance part-higher order modulation of subcarriers including 16 QAM and 64 QAM. Zhang Haodong finish the basic block diagram of the OFDM module and explore the impact of the subcarrier numbers on the OFDM system and then complete the advance part-image and video transmission.

Score	98