

# Design of 16-QAM Architecture for OFDM Transmitter in 4G Communication

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**Abstract**— Data on OFDM subcarriers is mapped (modulated) using common digital modulation schemes IEEE 802.11a/g WLANs uses BPSK, QPSK, 16-QAM, 64-QAM. Choice of QAM is to provide both phase and amplitude vary so we can use more number of sub channels in band. Also contain higher data rate than QPSK. Implementation of QAM is satisfies the specifications of the IEEE 802.11a/g. this architecture of QAM leads to power and area saving.

**Keywords**— QAM, OFDM, ASK, PSK

## I. INTRODUCTION

Audio data input interface hardware model for QAM system

Analog-To-Digital Conversion

A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, the tendency today is to change an analog signal to digital data. In this section we describe techniques, pulse code modulation. Fig.1 as shown below shows the system level model of the recorded voice signal to convert binary valves this valves are used for input for the QAM system.

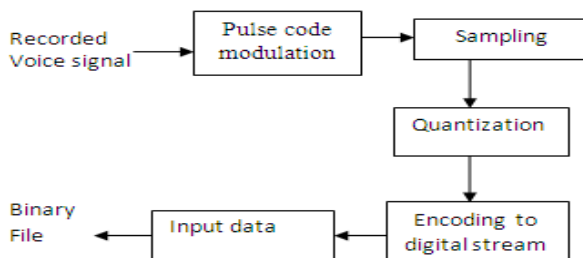


Figure 1: System level model

## II. PCM

PCM consists of three steps to digitize an analog signal:

1. Sampling
2. Quantization
3. Binary encoding

Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.

Filtering should ensure that we do not distort the signal, i.e. remove high frequency components that affect the signal shape. Figure 2 as shown below shows the components of PCM encoder

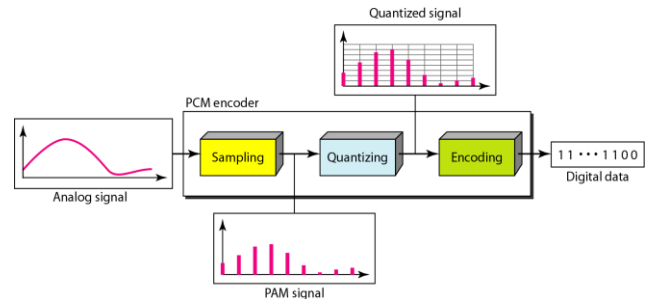


Figure 2: Components of PCM encoder

### A. Sampling

1.  $T_s$  is referred to as the sampling interval.
2.  $f_s = 1/T_s$  is called the sampling rate or sampling frequency.
3. There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - sample and hold, like natural but with single amplitude value
4. The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values
5. Analog signal is sampled every  $T_s$  secs.

Three different sampling methods for PCM

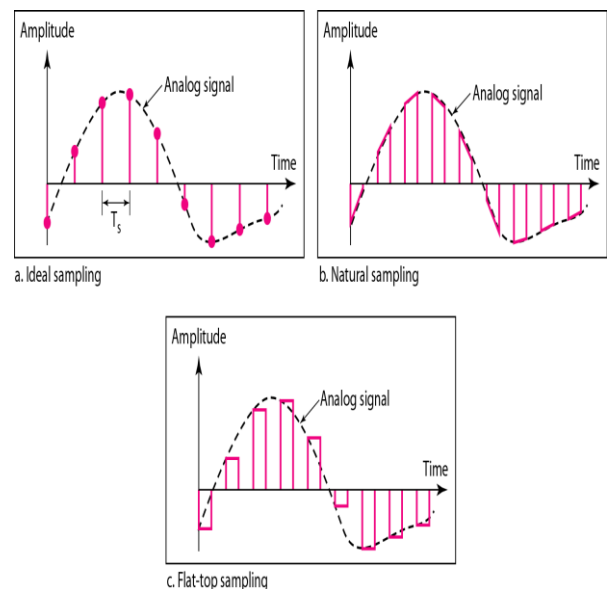


Figure 3: Three different sampling methods for PCM

The above fig.3 shows the three different sampling methods for pcm. According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal

**B. Quantization**

1. Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
2. The amplitude values are infinite between the two limits.
3. We need to map the *infinite* amplitude values onto a finite set of known values.
4. This is achieved by dividing the distance between min and max into L zones, each of height Δ.  
 $\Delta = (\text{max} - \text{min})/L$

**a. Quantization Levels**

1. The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
2. Each sample falling in a zone is then approximated to the value of the midpoint.

**b. Quantization Zones**

1. Assume we have a voltage signal with amplitudes  $V_{\text{min}}=-20V$  and  $V_{\text{max}}=+20V$ .
2. We want to use  $L=8$  quantization levels.
3. Zone width  $\Delta = (20 - -20)/8 = 5$
4. The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
5. The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5
6. Each zone is then assigned a binary code.
7. The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:  
 $n_b = \log_2 L$
8. Given our example,  $n_b = 3$
9. The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
10. Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

**c. Quantization and encoding of a sampled signal**

Figure 4 as shown below shows the quantization and encoding of our voice signal

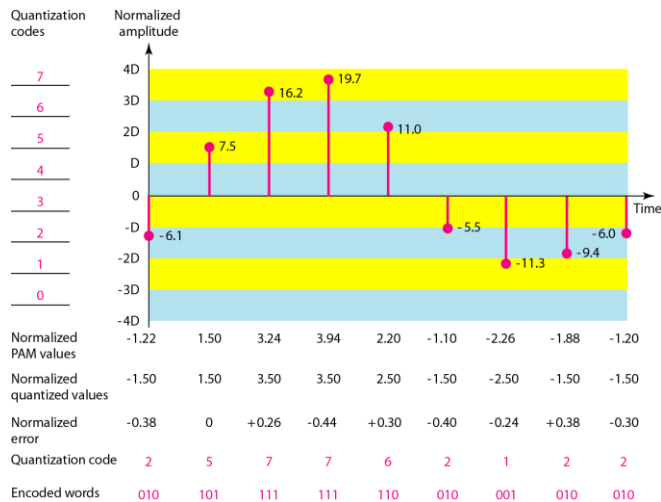


Figure 4: Quantization and encoding of a sampled signal

**d. Quantization Error**

1. When a signal is quantized, we introduce an error - the coded signal is an approximation of the actual amplitude value.

2. The difference between actual and coded value (midpoint) is referred to as the quantization error.
3. The more zones, the smaller Δ which results in smaller errors.
4. BUT, the more zones the more bits required to encode the samples -> higher bit rate

**III. DESIGN OF QAM**

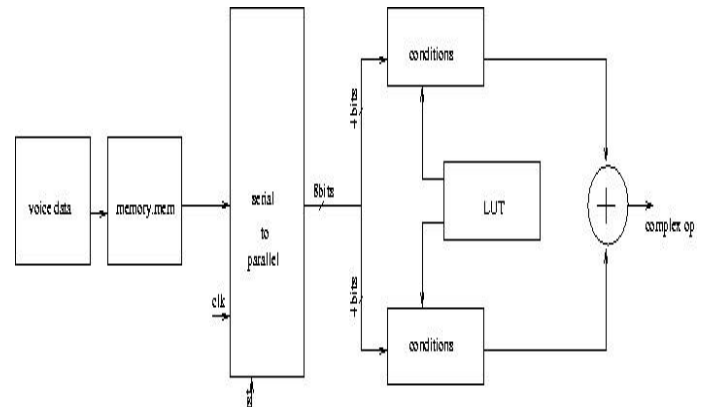


Figure 5. Design of QAM

The above figure 5 shows the design of Qam architecture. The recorded voice signal is convert to encoded binary valves this are stored in memory file the memory block is connected to serial to parallel converter to make it possible two carriers with the same frequency and phase in 90 degrees (sine and cosine) are modulated in amplitude by a multilevel signal (Figure 2), which through a multilevel binary converter generates the levels often a and b according to the number of binary input. For example, on a QAM modulator of two bits a one bit is sent to the converter in phase and the other bit is sent to the converter in quadrature.

Now for a modulator of four bits are sent to the converter in phase and the other four are sent to the converter in quadrature. Here we taken look up table for storing all possible 16 conditions instead of multipliers depending up on our input valves the complex output will appear on output side this will contains in phase and quadrature output.

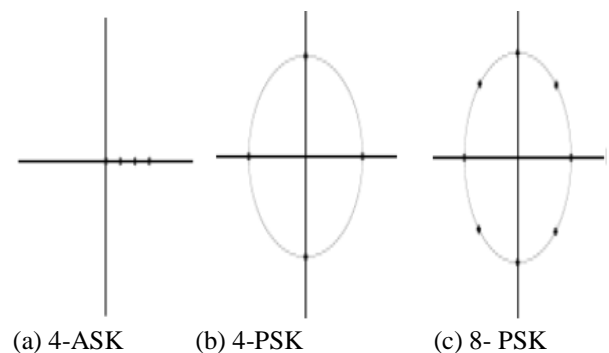


Figure 6: Diagrams constellation of ASK and PSK

**IV. MODULATION**

The result of the sum of these signals is a signal modulated both in phase and in quadrature As a and b there are some discrete levels, then we can give a diagram all possible combinations of a and b, called Diagram of constellation Figure.6 shows the diagram, constellation for modulation ASK and PSK.

The architecture diagram of a QAM modulator is mounted according to the discrete levels produced by the converter. Thus in a QAM modulator of two bits have only four points in the constellation, so this modulator is called as 4 QAM (Figure 6 (a)). Similarly in QAM modulator is of 4 bits and 16 QAM are so on. Note that there are several ways to implement the QAM constellation (Figure 6 (b) and Figure 6 (c)). The format of the constellation is determined by the conversion of angle.

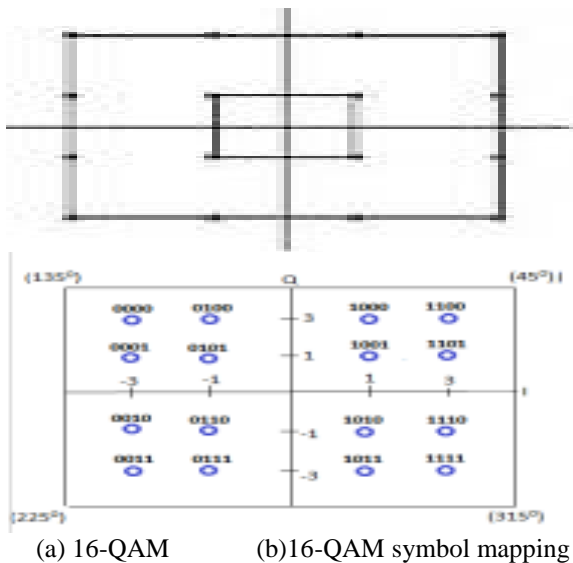


Figure 7: Constellation representation of QAM

**V. ENCODER'S CONSTELLATION**

The constellation maps of the encoder of the channel of Mn bits are at a point  $A_n + jB_n$  in the constellation diagram of the modulator. The decoder receives this point as how Mn Bits are transmitted. As said, it is important to note that this mapping made just one conversion from bits to the phase as represents, but it made no modulation. And need Specify how the constellation will be mapped, in order to implement this block However, regardless of the format of the constellation the block of the encoder can be done through a query to a table of conversion and implemented by the LUT that is in the ICs of FPGAs.

Table 1: 16-bit QAM Constellation

Symbol Transmitted	Carrier phase (In degrees)	Carrier amplitude
0000	135	1
0001	165	0.75
0010	195	0.75
0011	225	1
0100	105	0.75
0101	135	0.33
0110	225	0.33
0111	225	0.75
1000	75	0.75
1001	45	0.33
1010	315	0.33
1011	285	0.75
1100	45	1
1101	15	0.75
1110	345	0.75
1111	315	1

For example, a 16-QAM constellation, shown in Figure 7(a) and 16-QAM symbol mapping also shown in 7 fig(b) so that a and b are Binary numbers of 7 bits with different phase and angle are as shown in Table 1.

**VI. IMPLEMENTATION DETAIL AND RESULTS**

**Results**

**A. Relative signal strength of voice**

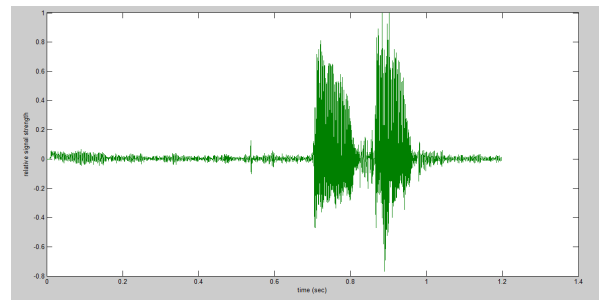


Figure 8: Relative signal strength of voice

The above figure 8 shows the relative signal strength of our voice signal simulated in MAT LAB

**B. Quantization of voice signal**

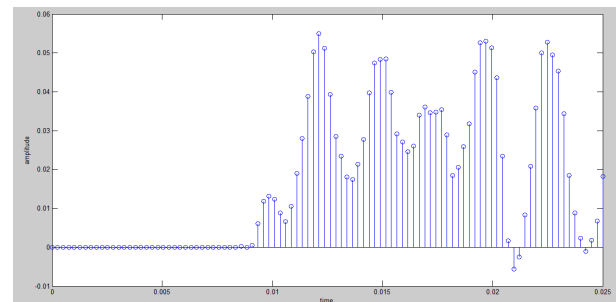


Figure 9: Quantization of voice signal

The above figure 9 shows Quantization of our voice signal simulated in MAT LAB here also shows the zones for taking the encoded binary values

**C. Encoding values stored in memory**

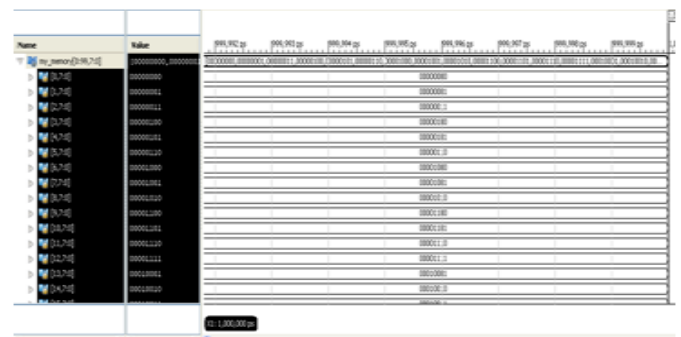


Figure 10: Encoding values stored in memory

The above figure 10 shows the encoded values are stored in memory(.mem) file using Xilinx 14.2

As an important part of the OFDM, modulation of signal is necessary requirement. So QAM is the modulation as well as demodulation techniques for the model. Apart from it there are other modulation techniques is also used like QPSK. Here we write the QAM code on VERILOG using Xilinx ISE 14.2 and Simulation result observe in Modelsim . Further results will be implemented it on the FPGA family Spartan Kit.

**D. Serial to parallel**

The serial-parallel converter of the transmitter has the function to receive data to be transmitted, assembles them in blocks of M bits, and then separates them into N channels, with Mn Bits each. Remember that the number of bits in each channel may be different.

**E. QAM (Quadrature Amplitude modulation)**

In Quadrature Amplitude modulation techniques here we use eight bit input as serially and it will transmitted to the same bit so it will generate the 4 point in each quadrant and 16 bit quadrature output with Different in phase and same in amplitude. As shown in result here it will give the 4 point output in each quadrat with same amplitude and different phase. In fig.11 and fig.12, Shows the RTL schematic of QAM and it gives the detailed of the various block and terminals of gate and flip-flops which is used to make LUTs and CLB, with Maximum Frequency: 134.012MHz.

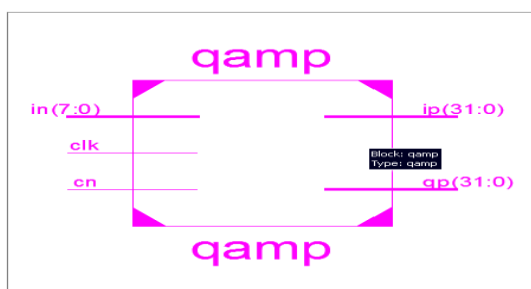


Figure 11: RTL schematic of QAM

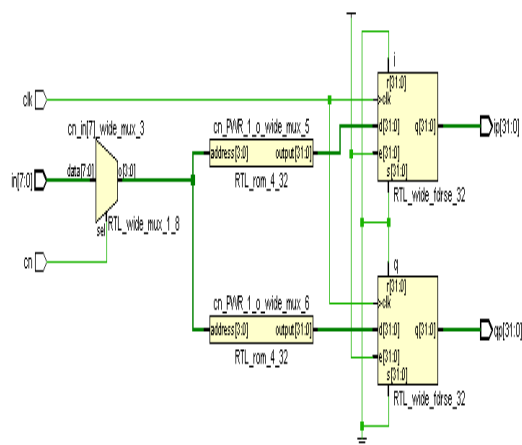


Figure 12: RTL Schematic of QAM architecture

Table 2: Design summary of QAM

Logic Utilization	Used	Available	Utilization
Number of slice LUTs	6	9112	0%
Number of fully used LUT-FF pairs	0	6	0%
Number of Bonded IOBs	74	232	31%
Number of BUFG/BUFGCTRLs	1	16	6%

**CONCLUSION**

16-QAM architecture is the most appropriate modulation scheme for OFDM based wireless broadband communication system because it contains higher data rate with less bandwidth and modeled with FFT/IFFT architecture instead of

bank of modulation. It also satisfies the specification of IEEE 802.11 a/g is described in this paper. And implementation of this architecture is advantages of technology mapping in terms of time area and power consumption.

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