Demonstration of Adaptable Quality Radio System for Broadcasting of Speech

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***Abstract*—This paper is presenting the design and implementation of a radio communication system for broadcasting of speech, with adaptable data rate. This system is to be seen as a "proof of concept", where the main goal is to demonstrate a radio system with feedback from receiver (RX) to transmitter (TX) such that the transmitted data rate adapts to the state of the radio channel, by evaluating the received bit error rate (BER). The data rate is varied by a factor 3 by switching between QPSK and QAM-64 modulation, while the bandwidth is fixed.**

I. Introduction

When designing any radio communication system, trade-offs has to be made between bandwidth, power, system complexity and bit rate. For a given bandwidth and transmit power, the bit rate could be varied be using different modulation schemes. As higher order modulation schemes require higher $\frac{E_b}{N_0}$, choosing optimum modulation scheme would require knowledge about the radio channel to maintain low enough BER at the receiver. This knowledge may be obtained by adding complexity in form of a feedback channel from receiver to transmitter. By performing some kind of error detection, the receiver may send information about detected error rates back to the transmitter. This enables system to adapt the data rate to the state of the radio channel, and thus provide better QoS for given power and bandwidth.

In this paper, we present a radio communication system with this kind of feedback structure. The system is to be seen as a "proof of concept" and the goal is not to propose a complete radio system for commercial use. As the main goal of the system is to demonstrate this feedback feature and all design choices are made with this in mind. We chose for instance not to implement any source encoding because it is not necessary to fulfil the purpose of our system, but would be highly favourable in a commercial system.

The adaptable quality is obtained by implementing a feedback path from the receiver to the transmitter. Figure 1 shows a top level block diagram of the proposed system. The figure shows that speech data is sent in the forward path from transmitter to receiver, and the number of detected errors is sent in the feedback path from receiver to transmitter. The forward and feedback paths will be referred to as the *data path* and the *BER path* respectively.

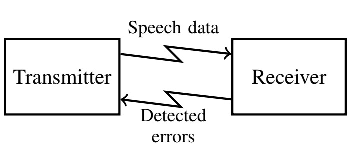

Fig. 1. Top level block diagram of proposed system. Speech data is sent in the forward path from transmitter to receiver and the number of detected errors is sent back from receiver to transmitter.

Throughout this paper, we will use the word *transmitter* when referring to the radio module that is transmitting the speech data and *receiver* when referring to the module that is receiving it. This is not to be confused with the terms TX and RX, which we use when referring to the transmit and receive port on each radio module.

The system is implemented using the software defined radio USRP-2901 [1] from National Instruments, which contains all necessary RF hardware. All the software parts of the system is implementing in C++ and is executed on a standard personal computer. Pre-written C-libraries are used for the parts of the system concerning interface to the USRP, the computer sound card etc. These parts will not be explained in detail, but references to the libraries will be given. For the remaining parts of the system, we focus on explaining the implementation on a behavioural level and detailed descriptions of C code implementations is avoided.

This paragraph will give tell the reader what to find in the remaining sections of the paper.

II. System Specifications

The proposed radio communication system switches between QPSK and QAM-64 modulation, at a fixed transmit power and bandwidth, in order to obtain adaptable sound quality. The system use the 2.4GHz ISM band with a carrier frequency of 2.415GHz. The system is designed for a transmission distance of 10 meter in an indoor environment with a bandwidth of 65kHz. Some key specifications of the system is listed in table I and II. Table I shows the parameters for the data path and the values are listed for low / high data rate transmission. Table II shows parameters for the simpler BER path.

TABLE I
System specifications - data path

System Variables	Value Low / High Data rate
Frequency f_0	2415 MHz
Modulation	QPSK/QAM-64
Bit per symbol m	$2/6$
Sound sampling rate f_s	$11025/22050$ Hz
Bits per sound sample b_s	$8/12$ bits
Sound datarate R_{ss}	$88.2/246.6$ kbits/s
Channel coding	Hamming (4,7)
Packet Parameters	
Packet header size	11 bits
Packet data length	128 symbols
Packet size	$256/768$ bits
Frame Parameters	
Training sequence type	Barker
Training sequence length	26 symbols
Training sequence size bits	$52/156$ bits
Frame size	$319/935$ bits
Burst Parameters	
Guard period	6 symbols
Burst size	$331/935$ bits
Transmission Characteristics	
System bit rate R_b	$191,35/577,18$ kbits/s
Symbol rate R_s	$95,67/96,20$ ksymbols/s
Pulse shaping filter	root raised cosine
Pulse shaping filter parameter α	0.3
Minimum signal bandwidth Δf	$62,2/62,5$ kHz

The burst format for the transmitted data is shown in figure 2. The bursts are different when using QPSK and QAM-64 modulation, because the same number of e.g. training symbols maps to a different number of bits. The data packages (a and b) is transmitted continuously with the indicated guard period. The BER packages is

TABLE II
SYSTEM SPECIFICATIONS - BER PATH

System Variables	**Value Low / High Data rate**
Frequency f_0	2420 MHz
Modulation	QPSK
Packet Parameters	
Packet header size	11 bits
Packet size	8 bits
Frame Parameters	
Training sequence type	Barker
Training sequence length	26 symbols
Training sequence size bits	52 bits
Frame size	71 bits
Transmission Characteristics	
System bit rate R_b	42.5 kbits/s
Symbol rate R_s	100 ksymbols/s
Pulse shaping filter	root raised cosine
Pulse shaping filter parameter α	0.3
Minimum signal bandwidth Δf	$62, 2/62, 5$ kHz

Training sequence	Header	Speech data	Guard bits
52	11	256	12

(a) QPSK data package

Training sequence	Header	Speech data	Guard bits
156	11	768	36

(b) QAM-64 data package

Training sequence	Header	BER data
52	11	8

(c) QPSK BER package

Fig. 2. Burst format for QPSK(a) and QAM-64(b) modulated data bursts, and QPSK modulated BER burst (c)

very small compared to the data packages, and is only transmitted ones per received data package. Thus no guard period is specified for these packages.

III. DESIGN DESCRIPTION

Figure 1 shows how the transmitter and receiver communicates within the system. Block diagrams for the two subsystems is shown in appendix A, figure 4 and 5. The behaviour will be explained in this section.

Because two different modulation schemes are used, the receiver need to know how to decode the incoming data packets. This problem is solved by using two different training sequences, both of length 26symbols. The training sequence is an appropriate repetition of barker sequences of length 7 and 13, for QPSK and QAM-64 modulation respectively. As shown in figure 4, after frame sync, the receiver perform a check on which barker sequence that was transmitted before de-mapping the symbols.

In the transmitter, a variable called *Session State* keep the information about what data quality and modulation scheme to use. As figure 4 indicates, the session state influences several blocks of the TX side in the transmitter. As shown in figure 5, the decision of when to change data quality is left completely to the transmitter. For every received data package, the receiver computes the number of detected errors and transmit this number back to the transmitter. As the figure indicates, these BER packages are always transmitted using QPSK-modulation and no FEC is used. Based on the received number of detected errors as well as previous error detections, the transmitter decides whether to change session state or not.

In the following subsections, all components of the data path and BER path systems will be described in detail.

A. Sound Producer and Sound Consumer

The blocks sound producer and sound consumer contains functionality for handling the sound input and output to the sound card of the computer. Sound producer reads sound samples at from the sound card at full quality, i.e. 16 bit, 44100 Hz stereo, and writes the samples to a queue accessible for the source encoder. Sound consumer equivalently reads sound samples from a queue controlled by the unpacking block and writes to the computer sound card.

B. Source Encoder/Decoder

The source encoder performs lossy compression of the produced sound samples. The samples produced by sound producer are stored as 64 bit unsigned integers (u64) even though the resolution is only 16 bits. This means that the 48 least significant bits of each sample is zero. Depending on the state of the system (High or low quality transmission) the source encoder read several u64s and pack them into one single u64. Five or eight samples are packed into each u64 depending on the state (see details in table I). In addition the sample rate are reduced by decimation with a factor of 2 or 4 for high and low quality respectively.

The source decoder performs the inverse operation, writing 2 or 4 copies of the same sample as u64's to the sound consumer.

C. Packing

Add header and write to packet queue. Need more info.

D. Forward Error Correction

The implemented FEC algorithm is Hamming (7,4). The implementation is a fast, pre-written C-code, written by Michael Dipperstein [2].

E. Symbol Mapping

The system uses Grey Code for mapping the binary data to a complex vector z. For a two bit sequence b_1b_0, with b_0 as the least significant bit, the QPSK mapping is obtained by equation 1 and 2.

$$\text{Re}[z_{qpsk}] = (-2b_0 + 1) \tag{1}$$
$$\text{Im}[z_{qpsk}] = (-2b_1 + 1) \tag{2}$$

For a 6 bit sequence $b_5b_4b_3b_2b_1b_0$ the mapping to QAM-64 is obtained by equation 3 and 4

$$\text{Re}[z_{qam64}] = (2b_5 - 1) \cdot [(-2b_3 + 1) \cdot (-2b_5 + 3) + 4] \tag{3}$$
$$\text{Im}[z_{qam64}] = (2b_6 - 1) \cdot [(-2b_4 + 1) \cdot (-2b_6 + 3) + 4] \tag{4}$$

The resulting constellations from these mappings are shown in figure 6 and 7 in appendix B.

F. Training sequence

The training sequence is added to the transmitted packet in the block *Add Barker* on figure 4. As previously described, the training sequence is a barker sequence of length 7 or 13 depending on the session state. In either case the total length of the training sequence is 26symbols, which means that the barker sequence is repeated to obtain the desired length.

G. Pulse shaping

The pulse shaping filter is applied in the last step before transmission, and is of the type Root Raised Cosine, with a Roll-off factor of 0.3. The same filter is applied as a matched filter in the first step after reception at the receive side.

H. Symbol synchronisation

Pulse align. Need info.

I. Frequency synchronisation

The implemented frequency synchronisation is a combination two algorithms; blind frequency search and the Mth-power algorithm. The blind frequency search algorithm is applied before frame synchronisation and provides a rough estimate of frequency offset. The purpose of this block is to ensure that the phase drift of the training symbols is small enough before frame synchronisation. The blind frequency search is done by applying a fixed set of frequency offsets to the input symbols and then computing the cross correlation between the input symbols and the two training sequences. The frequency offset yielding the highest value for the cross correlation gives the best estimate for the frequency offset. This way, the remaining frequency offset for the frame sync is ensured to be within an interval small enough for the frame synchronisation algorithm to work.

The Mth-power algorithm is applied after frame synchronisation and does both frequency and phase offset estimation. The algorithm operates on the training sequence only. The algorithm is based on mapping all symbols to the same point in the complex plane by raising each symbol to the 4th power. With out any phase or frequency shift, each symbol of the training sequence should be mapped to the point(-1, 0). The phase estimate of the acquired symbol is then $1/4$th of the angular deviation from this point. This is illustrated in figure 3. The frequency offset is estimated by tracking deviation in phase. The implemented Mth-power algorithm is executed in four stages.

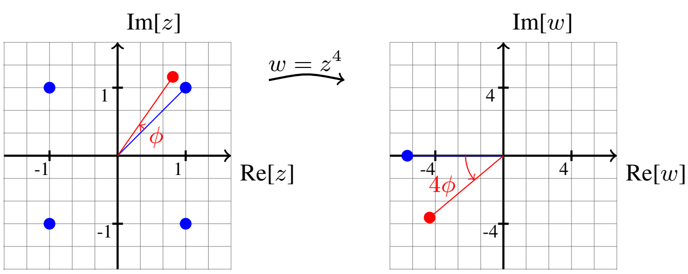

Fig. 3. Illustration the mapping $w = z^4$ which indicates how a phase error in QPSK modulated signal can be observed.

a) Estimate frequency offset by calculating the average phase difference between two adjacent symbols.
b) Estimate phase offset by first subtracting the estimated frequency offset from each sample, then calculate the average phase offset.
c) The above phase estimate is limited to $[0, \frac{\pi}{2}]$, so the phase offset need to be moved to the correct quadrant. Because the first symbol in the training sequence is always $1 + i$, an angle of $\pi/2$ is added to the the phase estimate until the first symbol is mapped to the first quadrant.
d) Apply the estimated phase and frequency to all remaining symbols.

J. Frame synchronisation

Frame synchronisation is done by computing the crosscorrelation between the two training sequences and the received symbols. Value of the crosscorrelation is compared to a pre set threshold of 13. The first index that gives a value exceeding this threshold is considered to be the beginning of the frame. NEED MORE INPUT

IV. Link Budget

The link budget for the system is shown in table III. As the purpose of the system is to demonstrate the feedback feature, the system is designed for the test environment only, which is reflected in the link budget. The system is designed to operate indoors with a distance of 10 meters between transmitter and receiver. Table III shows losses from path loss, loss in TX and RX and some estimated key parameters at the receiver, such as E_b/N_0 and BER.

TABLE III
Link Budget

TX Loss	**Value Low / High Data rate**
PA Power, P_{PA}	10dBm
TX Connector Loss, L_{ConT}	-0.3dB
TX Power, P_T	9.4dBm
TX Antenna Gain, G_T	3dBi
Effective (Isotropic) Radiated Power, EIRP	12.4dBm
Path Loss	
Distance, d	10m
Floor loss factor, $Pf(n)$	0dB
Distance power loss coefficient, N	38
Total ITU path loss, L_P	-77.66dB
RX Loss	
RX antenna gain, G_R	43dBi
RX connector loss, L_{ConR}	-0.3dB
Total RX Loss, L_R	2.4dB
Total Received Power, P_R	-62.86dBm
RX Loss	
Antenna Noise Density, N_0	-145.73dbm/Hz
Antenna Total Noise Power, N	-97.806dBm
RX Noise Figure, NF	7.0dB
Small Scale fading margin, M_{ssf}	9.4dB
RX Properties	
Carrier-to-noise ratio, C/N	18.548dB
Eb over N0, E_b/N_0	13.667dB/8.893dB
Eb over N0, E_b/N_0	23.262/7.749
Bit error rate, BER	$8.56 \cdot 10^{-08}/3.17 \cdot 10^{-04}$

As the system is designed to switch between two different modulation formats the received E_b/N_0should not be carefully tuned. The link budget is designed such that the $E_b/N_0$is mostly good enough for QAM-64 modulation under line of sight (LOS), but forces the system to switch to QPSK if the LOS is lost.

The different parts of the link budget will be discussed in this section.

A. TX and RX Loss

The value for connector loss is taken from datasheets of standard coaxial RF connectors [3]. The antenna gain value is taken from the data sheet [4] which reports a peak gain of 3.4 dBi. We used the value 3 dBi in the link budget to account for suboptimal conditions. The launch power, PA Power, was adjusted after measurements to obtain appropriate E_b/N_0 at the receiver.

B. Path Loss

The estimated path loss constitutes solely of the propagation loss obtained from the ITU Indoor Propagations Loss Model [5]. The loss model consists of two adjustable factors, the distance power loss coefficient, N, and the floor loss penetration factor, $P_f(n)$. The latter is set to 0, and the former was set to 38 after calibrating the test environment. Other loss factors such as pointing loss and polarisation loss was considered but measurements showed that the amount of reflections in the room made pointing and polarisation irrelevant to the received power. More details on the measurements is given in section VI.

C. RX Noise

The antenna noise density was measured with a spectrum analyser and the estimated value was taken as an average of several single runs. The noise figure of the receiver is included to account for noise added by the radio hardware, with value taken from the data sheet. The small scale fading margin, M_{ssf}, is included to account for variations in received power. This margin was obtained by evaluating several measurements of received power using a spectrum analyser. The particular value is taken to be two times the standard deviation of the measured values.

D. RX Properties

Some key properties of the received signal is calculated based on the estimated values in the link budget. The BER and E_b/N_0 is calculated for the two modulation schemes separately. The bit error rate is calculated for QPSK and QAM-64 by equation 5 and 6 respectively.

$$P_B \approx \frac{1}{2}\,\mathrm{erfc}\sqrt{\frac{E_b}{N_0}} \tag{5}$$

$$P_B \approx \frac{2}{\log_2 M}\left(1-\frac{1}{M}\right)\mathrm{erfc}\left(\sqrt{\frac{3\log_2 M}{2(M-1)}\cdot\frac{E_b}{N_0}}\right) \tag{6}$$

V. Design Motivation

The goal of the proposed system is to demonstrate a radio communication system with adaptable sound quality. This main goal is the background for all design choices that is made. In this system, the motivation behind the solutions described in section III is given.

A. Choice of modulation scheme

The system uses QPSK for low data rate transmission and QAM-64 for high data rate transmission.

B. Source Encoder/Decoder

We chose not to implement any source encoding except for the primitive reduction of data rate. A source encoder removes redundancy in the speech data and increase the information content of every transmitted symbol. This functionality is not necessary to demonstrate the adaptable quality and is therefore not implemented.

The source encoder also makes the system more sensitive to bit errors, as the information content in each bit is increased. Our system is designed to operate at E_b/N_0 low enough to demonstrate the quality adaption. When the received E_b/N_0 falls low enough to make the system change from QAM-64 to QPSK modulation, we would like the system to have a large margin before the BER for QPSK also get too high and the link is broken. Using a source encoder would make this window smaller and the system would be harder to demonstrate.

VI. Measurements and Verification

References

[1] National Instruments, "Usrp-2901 specifications - national instruments," https://www.ni.com/pdf/manuals/374925c.pdf, (Accessed on 04/16/2020).
[2] M. Dipperstein, "Hamming (7,4) code discussion and implementation," http://michael.dipperstein.com/hamming/, (Accessed on 04/20/2020).
[3] TE Connectivity, "Rf coaxial connectors," https://www.mouser.com/datasheet/2/418/NG_DS_1-1773725-8_RF_COAX_QRG_0114_TE-1948_RFcoaxi-1232379.pdf, (Accessed on 04/14/2020).
[4] Siretta, "Delta-7a datasheet," https://www.siretta.com/wp-content/uploads/2020/01/Delta-7A-Rev-1.7.pdf, (Accessed on 04/14/2020).
[5] International Telecommunication Union, "Recommendation itu-r p.1238-10," https://www.itu.int/dms_pubrec/itu-r/rec/p/R-REC-P.1238-10-201908-I!!PDF-E.pdf, (Accessed on 04/14/2020).

Appendix A
Block Diagram

Appendix B
Symbol Mapping Scheme

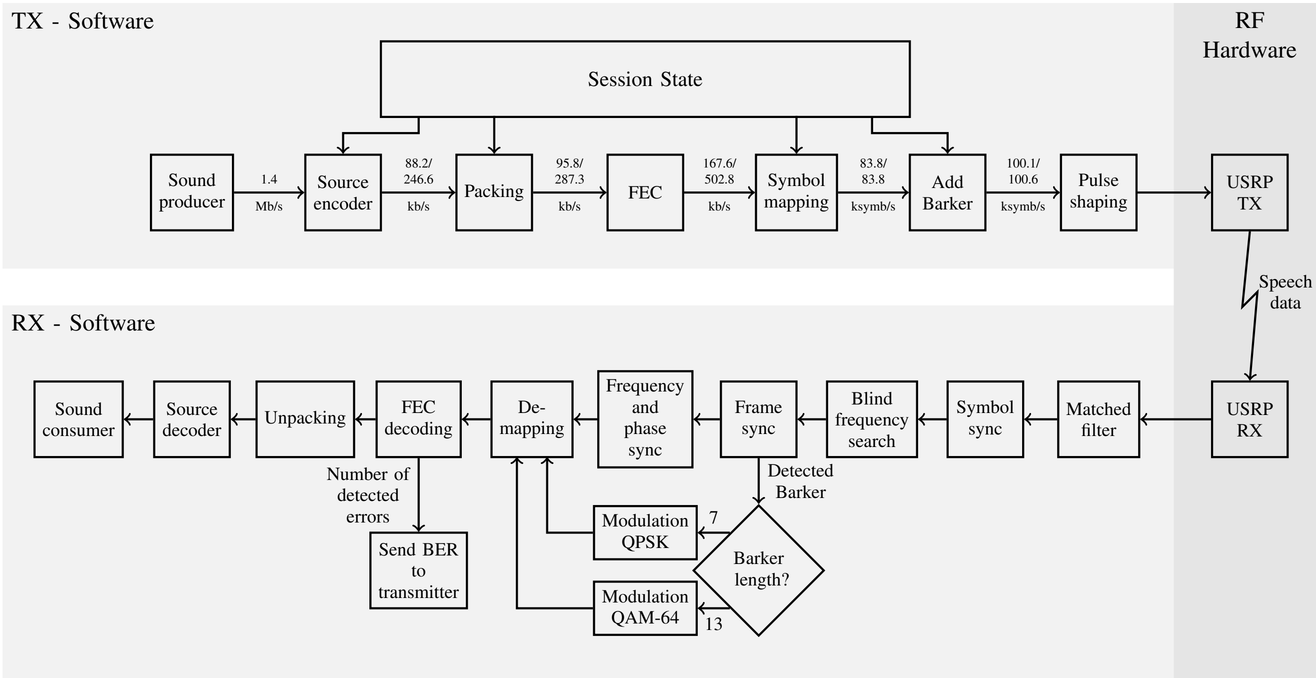

Fig. 4. Block diagram of data packet system. This block diagram shows the forward path of the system, where speech data is being transmitted.

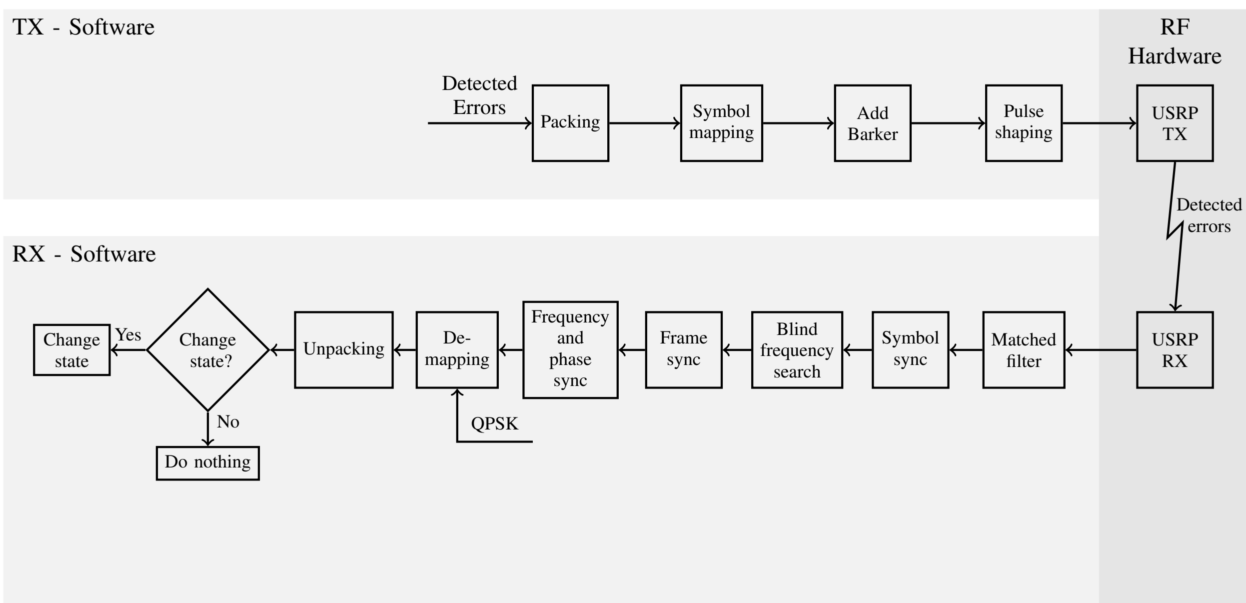

Fig. 5. Block diagram of BER packet system. This sub-system constitutes the feedback path where the receiver transmit information about detected error rate back to the transmitter.

```
%====QPSK=====%
%=============%
%   Coding    %
%  01  |  00  %
% ----------- %
%  11  |  10  %
%=============%
```

Fig. 6. Constellation mapping for QPSK modulated symbols

```
%==========================64 QAM==========================%
%==========================================================%
%                           Coding                         %
%  010000 010010 011010 011000 |111000 111010 110010 110000  %
%                              |                           %
%  010001 010011 011011 011001 |111001 111011 110011 110001  %
%                              |                           %
%  010101 010111 011111 011101 |111101 111111 110111 110101  %
%                              |                           %
%  010100 010110 011110 011100 |111100 111110 110110 110100  %
%  ----------------------------|--------------------------   %
%  000100 000110 001110 001100 |101100 101110 100110 100100  %
%                              |                           %
%  000101 000111 001111 001101 |101101 101111 100111 100101  %
%                              |                           %
%  000001 000011 001011 001001 |101001 101011 100011 100001  %
%                              |                           %
%  000000 000010 001010 001000 |101000 101010 100010 100000  %
%==========================================================%
```

Fig. 7. Constellation mapping for QAM-64 modulated symbols