

# 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. The data rate is varied by a factor 2 by switching between QPSK and QAM-16 modulation, while bandwidth and transmit power is fixed. The proposed system is implemented with a -60dBc bandwidth of 232kHz and a transmit power of -10dBm.

The adaptive quality feature is verified and the system changes quality immediately when the detected error rate drops below a predefined threshold. The measured bit error rates are  $6.0 \cdot 10^{-2}$  and  $1.4 \cdot 10^{-4}$  for high and low data rates respectively. The total delay from transmitter to receiver is measured to 5ms, making the system well suited for two-way communication as well as broadcasting.

## 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 by 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 in order 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 the 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. The adaptable quality is obtained by implementing a feedback path from the receiver to the transmitter, using frequency-division duplexing (FDD). 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. .

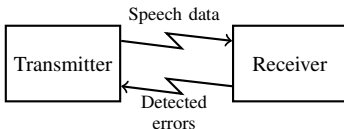


Fig. 1. Top level block diagram of proposed system. Speech data is sent in the forward path from transmitter to receiver and the number of detected errors is sent back from receiver to transmitter.

The proposed system is designed to switch between two different data rates, using two different modulation formats. Figure 2 shows a qualitative illustration of the adaptive quality concept. The proposed system may be further improved by adding more levels, yielding even better utilization of available resources.

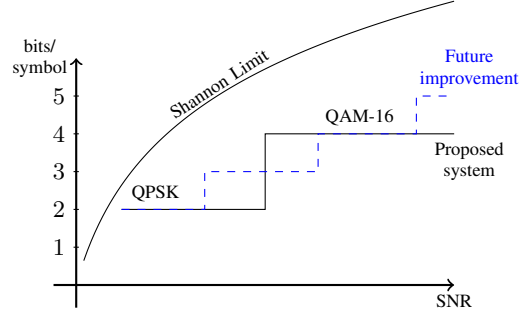


Fig. 2. Qualitative illustration of the adaptive quality concept

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.

In section II we present the system specifications and the link budget is presented in section III together with associated measurements. A detailed description of the proposed system is given in section IV and the motivation behind some crucial design decisions is presented in section V. Performed measurements is presented in section VI before a final conclusion is given in section VII.

## II. SYSTEM SPECIFICATIONS

The proposed radio communication system switches between QPSK and QAM-16 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 and 2.455GHz for the data path and BER path respectively. The system is designed for a transmission distance of 5 meter in an indoor environment.

Some key system specifications are 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 simplest BER path.

The burst format for the transmitted data is shown in figure 3. The bursts are different when using QPSK and QAM-16 modulation, because the same number of training symbols maps to a different number of bits. The data packets (a and b) is transmitted continuously with the indicated guard period. The BER packets is very small

TABLE I  
SYSTEM SPECIFICATIONS - DATA PATH

System Variables	Value Low / High Data rate
Frequency $f_0$	2415 MHz
Modulation	QPSK/QAM-16
Sound sampling rate $f_s$	11025/22050 Hz
Bits per sound sample $b_s$	12 bits
Sound datarate $R_{ss}$	132/265 kbits/s
Channel coding	Hamming (4,7)
<b>Packet Parameters</b>	
Packet header size	8 bits
Packet size	512 bits
<b>Frame Parameters</b>	
Training sequence type	Barker
Training sequence length	26 symbols
<b>Burst Parameters</b>	
Guard period	2 symbols
Burst Length	484/256 symbols
<b>Transmission Characteristics</b>	
Symbol rate $R_s$	150 ksymbols/s
Samples per Symbol	8
Pulse shaping filter	root raised cosine
Pulse shaping filter parameter $\alpha$	0.5
Minimum Nyquist bandwidth $\Delta f$	112,5 kHz

TABLE II  
SYSTEM SPECIFICATIONS - BER PATH

System Variables	Value Low / High Data rate
Frequency $f_0$	2455 MHz
Modulation	QPSK
Channel coding	Hamming (4,7)
<b>Packet Parameters</b>	
Packet header size	8 bits
Packet size	26 bits
<b>Frame Parameters</b>	
Training sequence type	Barker
Training sequence length	26 symbols
<b>Burst Parameters</b>	
Burst Length	43 symbols
<b>Transmission Characteristics</b>	
Symbol rate $R_s$	150 ksymbols/s
Samples per Symbol	8
Pulse shaping filter	root raised cosine
Pulse shaping filter parameter $\alpha$	0.5
Minimum Nyquist bandwidth $\Delta f$	112,5 kHz

compared to the data packets, and is only transmitted ones per received data packet. Thus no guard period is specified for these packets.

### III. 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 2 meters between transmitter and receiver. Table III shows losses from propagation, loss in TX and RX and some estimated key parameters at the receiver.

The value for connector loss is taken from datasheets of standard coaxial RF connectors [2]. The antenna gain value is taken from the data sheet [3] 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,  $P_{PA}$ , was adjusted after measurements to obtain appropriate  $E_b/N_0$  at the receiver.

The estimated path loss constitutes solely of the propagation loss obtained from the ITU Indoor Propagations Loss Model [4]. The

Training sequence 52	Session ID 3	Packet ID 5	Payload 512	Guard bits 4
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(a) QPSK data packet

Training sequence 104	Session ID 3	Packet ID 5	Payload 512	Guard bits 8
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(b) QAM-16 data packet

Training sequence 52	Session ID 3	Packet ID 5	Payload 26	
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(c) QPSK BER packet

Fig. 3. Burst format (bits) for QPSK(a) and QAM-16(b) modulated data packets, and QPSK modulated BER packet (c)

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, $P_T$	-10.6dBm
TX Antenna Gain, $G_T$	3dBi
Effective (Isotropic) Radiated Power, EIRP	-7.6dBm
<b>Path Loss</b>	
Distance, $d$	5m
Floor loss factor, $Pf(n)$	0dB
Distance power loss coefficient, $N$	38
Total ITU path loss, $L_P$	-66.22dB
<b>RX Loss</b>	
RX antenna gain, $G_R$	3dBi
RX connector loss, $L_{ConR}$	-0.3dB
Total RX Loss, $L_R$	2.4dB
<b>RX Properties</b>	
Total Received Power, $P_R$	-71.42dBm
Antenna Noise Density, $N_0$	-145.73dBm/Hz
Antenna Total Noise Power, $N$	-92.72dBm
RX Noise Figure, $NF$	7.0dB
<b>RX Properties</b>	
Carrier-to-noise ratio, $C/N$	14.3dB
$E_b$ over $N_0$ , $E_b/N_0$	12.54dB/9.53dB

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. Figure 4 shows the measured path loss and the prediction from the ITU model before and after adjusting the power loss coefficient.

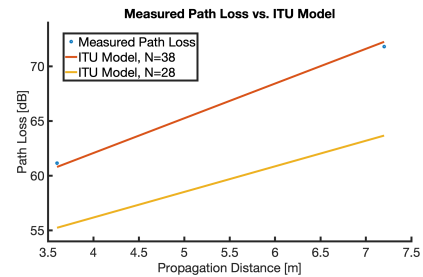


Fig. 4. Measured path loss vs. estimated path loss from the ITU Indoor Propagations Loss Model, before and after adjusting the power loss coefficient

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.

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.

#### IV. DESIGN DESCRIPTION

Figure 1 shows how the transmitter and receiver communicates within the system. Block diagrams for the two subsystems are shown in appendix A, figure 11 and 12. The data rate between each block of the data path is indicated with the thin arrows. The behaviour of the system 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 in the beginning of each frame. As shown in figure 11, after frame sync, the receiver perform a check on the received Barker sequence 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 11 indicates, the session state influences several blocks of the TX side of the transmitter. The decision of when to change data quality is left completely to the transmitter. For every received data packet, the receiver computes the number of detected errors and transmit this number back to the transmitter. These BER packets are always transmitted using QPSK-modulation. Based on the received number of detected errors the transmitter decides whether to change session state or not.

The sound producer and sound consumer contains functionality for handling the sound input and output to the sound card of the computer. They are implemented using the Windows API [5]. Sound producer reads sound samples from the sound card at full quality (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.

The source encoder performs lossy compression of the produced sound samples. The bit resolution is reduced to 12 bit per sound sample, and the sampling rate is reduced by a factor 2 or 4 depending on the session state. The source decoder performs the inverse operation, writing 2 or 4 copies of the same sample to the sound consumer.

In the packing block, sound data is read from the source encoder, a header is added, and the packet is sent to the packet queue. The eight bit packet header consist of a three bit session ID and a five bit packet ID.

A scrambler is implemented before FEC, which computes a bitwise XOR between a pseudo random bit string and the packet. The bit string used for scrambling is of the same size as the packet itself. The scrambler performs the exact same operation at RX and TX.

The implemented FEC algorithm is Hamming (7,4). The implementation is a fast, pre-written C-code, written by Michael Dipperstein [6].

The system uses Grey Code for mapping the binary data to a complex vector  $z$ . The mapping schemes are shown in figure 5.

The training sequence is added to both I and Q symbols in the block *Add Barker*. The training sequence is an appropriate repetition of Barker sequences of length 7 and 13 for QPSK and QAM-16 modulation respectively. The total length of the training sequence is 26 symbols.

In the last step before transmission, the symbols are upsampled by a factor 8 and filtered with a pulse shaping filter. The filter is Root Raised Cosine with a roll-off factor of 0.3. The same filter is

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

(a)

%=====16QAM=====			
%=====			
% Coding %			
% 0000	0100		1100 1000
%   %			
% 0001	0101		1101 1001
% ----- %			
% 0011	0111		1111 1011
%   %			
% 0010	0110		1110 1010
%=====			

(b)

Fig. 5. Symbol mapping for QPSK (a) and QAM-16 (b) modulated symbols

applied as a matched filter in the first step at the receive side before the samples are downsampled again.

The USRP is configured to transmit continuously with a symbol rate of 150 symbols per second and transmit arrays of zeros when no data is available. The USRP interface is implemented using the NI-USRP DLL [7].

Symbol synchronisation is done by choosing the sample offset that maximizes the signal energy.

Frame synchronisation is done by computing the crosscorrelation between the two training sequences and the received symbols. The value of the crosscorrelation is compared to a pre set threshold. The first index that gives a value exceeding this threshold is considered to be the beginning of the frame. After a frame is found, only the frame symbols are passed along to the next block.

##### A. Frequency and Phase Synchronisation

The implemented frequency synchronisation algorithm is based on the Mth-power algorithm, which is further improved by using a Kalman filter. Frequency offset is estimated from the training sequence only. The Mth-power algorithm estimates the phase offset of MPSK modulated symbols, by mapping all symbols to the same point in the complex plane. By tracking the angular difference between consecutive symbols, the frequency offset may be estimated. In our case, the QPSK symbols of the training sequence is raised to the 4th power, and the phase is estimated as indicated in figure 6.

For each received frame, the frequency offset is estimated by the mean of the angular difference between all consecutive symbols of the training sequence. The estimate obtained from a single frame at time  $k$  is referred to as  $\tilde{\omega}_k$ .

The accuracy of the estimate is improved by using a Kalman filter. The state of the Kalman filter is the one-dimensional state vector  $\omega_k$  which is the true frequency offset between the transmitter and the receiver at time  $k$  (i.e. frame  $k$ ). The state is represented by the *a posteriori* state and variance estimate  $\hat{\omega}_{k|k}$  and  $\hat{\sigma}_{k|k}$ , where the subscript  $n|m$  indicates the estimate of time  $n$  given observations up to and including time  $m \leq n$ .

The estimate is obtained in two stages, referred to as the *prediction stage* and the *update stage*. In the prediction stage, the *a priori*

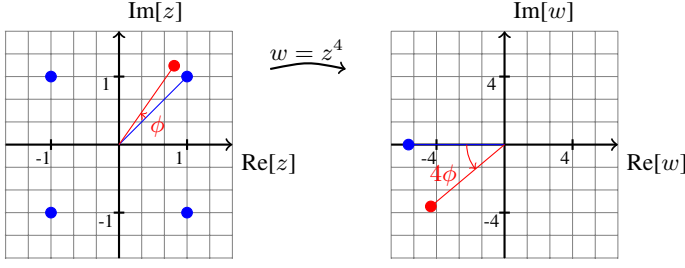


Fig. 6. Illustration the mapping  $w = z^4$  which indicates how a phase error in QPSK modulated signal can be observed.

estimates  $\hat{\omega}_{k|k-1}$  and  $\hat{\sigma}_{k|k-1}$  is obtained by equations 1 and 2.  $\sigma_f$  is the stationary variance of the process noise and must be tuned carefully in order to obtain appropriate convergence speed. In our system  $\sigma_f$  was set to  $0.8 \cdot 10^{-6}$ .

$$\hat{\omega}_{k|k-1} = \hat{\omega}_{k-1|k-1} \quad (1)$$

$$\hat{\sigma}_{k|k-1} = \hat{\sigma}_{k-1|k-1} + \sigma_f \quad (2)$$

In the update stage, the *a posteriori* estimates  $\hat{\omega}_{k|k}$  and  $\hat{\sigma}_{k|k}$  is obtained by equations 3 and 4.  $\tilde{\omega}_k$  and  $\tilde{\sigma}_k$  is the observed frequency offset and estimated observation noise at time  $k$  respectively.

$$\hat{\omega}_{k|k} = \hat{\omega}_{k|k-1} + \frac{\hat{\sigma}_{k|k-1}}{\hat{\sigma}_{k|k-1} + \tilde{\sigma}_k} (\tilde{\omega}_k - \hat{\omega}_{k|k-1}) \quad (3)$$

$$\hat{\sigma}_{k|k} = \hat{\sigma}_{k|k-1} \frac{\tilde{\sigma}_k}{\tilde{\sigma}_k + \hat{\sigma}_{k|k-1}} \quad (4)$$

After the estimates  $\hat{\omega}_{k|k}$  and  $\hat{\sigma}_{k|k}$  is obtained,  $\hat{\omega}_{k|k}$  is applied to each symbol in frame  $k$ . The phase offset is then estimated as the mean of the angular deviation between the training sequence of the received symbols, and the ideal Barker sequence.

## 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 section, the motivation behind the key specifications and some of the solutions described in section IV is given.

### A. System specifications

1) *Transmit Power*: As shown in table III, the transmitter PA power,  $P_{PA}$ , is set to -10dBm. As the purpose of the system is to demonstrate the adaptable quality feature, the transmit power was tuned to a value suitable for this purpose. Too high transmit power would give a received  $E_b/N_0$  high enough to always transmit at the best quality, thus disabling us from demonstrating the quality adaption.

2) *Modulation Scheme*: The system uses QPSK for low data rate transmission and QAM-16 for high data rate transmission. QPSK has the advantage of constant symbol amplitude and therefore reduces nonlinear effects in the amplifiers. Because of the orthogonality between the I and Q signals, QPSK also gives twice the capacity as BPSK for same  $E_b/N_0$  sensitivity. QAM-16 is chosen for high data rates because the information content in each symbol is twice as big as for QPSK, making the difference big enough to demonstrate an audible effect.

3) *Bandwidth*: The motivation behind the adaptable quality feature, is to maximise data rate, for fixed bandwidth and transmit power. To demonstrate this feature, the system is designed with a fixed bandwidth, equal for both quality levels. The specific value for the bandwidth is however not important for this demonstration. The transmitted symbol rate is chosen large enough to enable both QPSK and QAM-16 transmission. The ideal Nyquist bandwidth is calculated from the symbol rate and the properties of the pulse shaping filter, and presented in table III.

### B. System Architecture

The adaptable quality feature requires some additional logic, compared to a system without a feedback path. One important design choice is to determine where the decision of changing quality level (referred to as *state*) is to be made. During the design of the proposed system, two solutions were considered:

- 1) The receiver decides when to change state and sends a simple message to the transmitter, telling it to change state. The receiver has its own state variable, and updates it after sending the message to the transmitter. The advantage of this solution is that the receiver always know how the received symbols are modulated. Thus, no state information is needed in the transmitted packet. The draw back is that there will be some delay from the receiver ask for a change of state, until state change is completed. This will either cause a few packets to get lost in the meantime, or extra logic must be implemented to handle the issue. Extra logic must also be implemented to handle the case of bit errors in the “change-state-message”.
- 2) The transmitter decides when to change state and the receiver continuously send information about detected error rate to the transmitter. This requires information about the state to be contained in the data packet and the receiver will need more complexity in order to determine the incoming modulation format. However, this will eliminate the senario when the transmitter and the receiver are in different states.

We chose to implement the second solution. The state information is kept in the two different Barker sequences. Because we would have to search for one Barker sequence in the frame synchronisation anyway, searching for one more would be both computationally fast and easy to implement. In addition, the good autocorrelation properties of the Barker sequence makes the state information robust against noise, and no extra FEC is needed to keep this information safe.

### C. Implemented Functionality

1) *Source Encoding*: 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-16 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.

2) *Scrambler*: Analysis of the received constellation diagram clearly showed that some symbols occurred more frequently at the receiver than others. Both the RF hardware and the pulse align algorithm are highly sensitive unequal energy distribution between I and Q. When comparing the received constellation from speech data to the constellation from random data, we could see that the quality was considerably increased when transmitting white data. We therefore implemented a scrambler to whiten the speech data before transmission.

3) *FEC*: The implemented FEC algorithm is Hamming(7,4) and serves two purposes. In the data path, the receiver must estimate the number of bit errors and send this number back to the transmitter. In the BER path, the transmitter use the FEC to determine if the received BER packet is valid or not. If the number of bit errors in the BER packet is above a predefined threshold, the packet is considered as invalid in order to reduce the risk of undesired changes of quality level.

This particular algorithm was chosen for its simplicity and decent trade-off between redundancy overhead and minimal Hamming distance. The algorithm allows for detection of any two-bit errors in every 7 bit code word, which was considered sufficient for this purpose. In addition, pre-written C-code implementation exists [6] which saved development time. More advanced FEC algorithms should be considered for applications where bandwidth limitations require less overhead, or error correction is necessary.

4) *Frequency and Phase Synchronisation*: The Mth-power algorithm is chosen for frequency synchronisation because of the good accuracy and simple implementation. By estimating the frequency offset as the mean of the angular deviation between all consecutive symbols, the ML estimate of the frequency offset is obtained. In addition, the Mth-power algorithm does not require a known training sequence, and may be applied to all data samples when QPSK modulation is being used. By extending the algorithm to a Kalman filter, the mean square error between the estimated offset and the actual offset is minimised. The improvement from the Kalman filter is clearly seen in figure 7 which shows the estimated frequency offset at the receiver with and without Kalman filter.

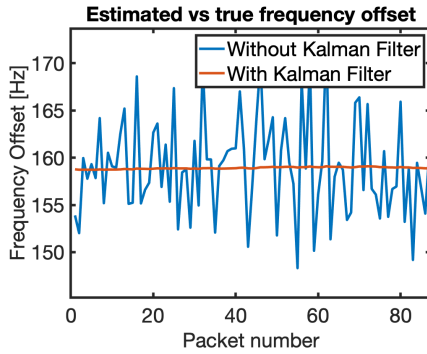


Fig. 7. Estimated frequency offset with and without Kalman filter

The Mth-power algorithm is however not linear, making it less suited for phase estimation. Therefore, the linear algorithm based on comparing the phase directly to the training sequence is used for estimating the phase.

## VI. MEASUREMENTS AND VERIFICATION

In this section we present measurements and verifications of the proposed system. We first present a verification of the system specifications, such as bandwidth and power, in section VI-A. In section VI-B the system performance will be evaluated under different conditions, and the key parameters BER and EVM will be presented.

### A. Verification of System Parameters

The verified system specifications is summarised in table IV.

TABLE IV  
SUMMARISED MEASUREMENTS OF SYSTEM PARAMETERS

System Parameter	Measured Value
Half-Power Bandwidth	232kHz
PA power, $P_{PA}$	-10dBm
Delay	5ms

The signal bandwidth was analyzed using MATLAB's signal analyzer toolbox [8]. The resulting power spectrum is shown in figure 8. The -60dBc bandwidth is measured to 232kHz. The transmit power was verified by first calibrating the USRP power spectral density using a spectrum analyser, and then calculating the total transmit power from measured bandwidth<sup>1</sup>. This way, the transmit power was measured -10dBm.

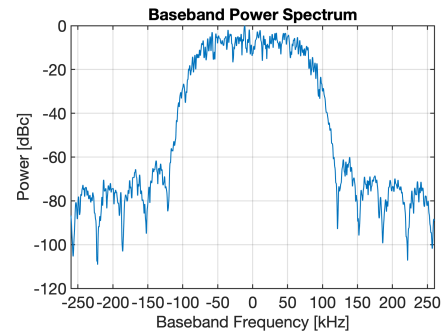


Fig. 8. Received signal power spectrum

The system delay was measured by transmitting a known bit sequence and evaluating the time delay from sound producer to sound consumer. The delay was measured by running both the transmitter and receiver software on the same computer. The average measured delay was 5ms with a sample standard deviation of 2ms.

### B. Performance Measurements

The system is tested in an indoor environment with both USRP's at the same altitude, 1 meter apart and no polarization mismatch<sup>2</sup>. As the system is designed to change quality level depending on the state of the radio channel, different radio channels should be used when verifying the system performance. However, for the sake of reproducibility, we chose to vary the received SNR by adjusting the transmit power instead of changing the radio channel. Two different transmit power levels were used to verify transmission at both qualities. High quality transmission was measured with a transmit power of -10dBm and low quality with -25dBm.

Eye diagram and constellation diagram is provided for both transmit power levels and both modulation formats, together with some key performance measurements. For each transmit power level, the BER is calculated by transmission of a known bit sequence and the average EVM is calculated from constellation diagram, normalised to the peak signal power.

The diagrams for high and low transmit power are summarised in figure 9 and 10 respectively. Calculated performance parameters are summarised in table V and VI for high and low transmit power respectively.

<sup>1</sup>Due to the COVID-19 situation the lab equipment was not available for measurements on the implemented system

<sup>2</sup>The measurements had to be performed in a tiny student apartment because the lab was closed



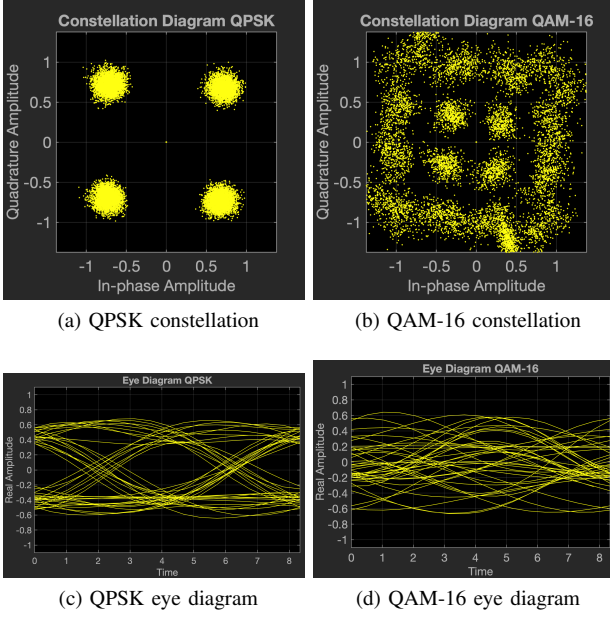


Fig. 9. Eye diagram and constellation for QPSK (a and c) and QAM-16 (b and d) modulated symbols. Transmit power:  $-10\text{dBm}$

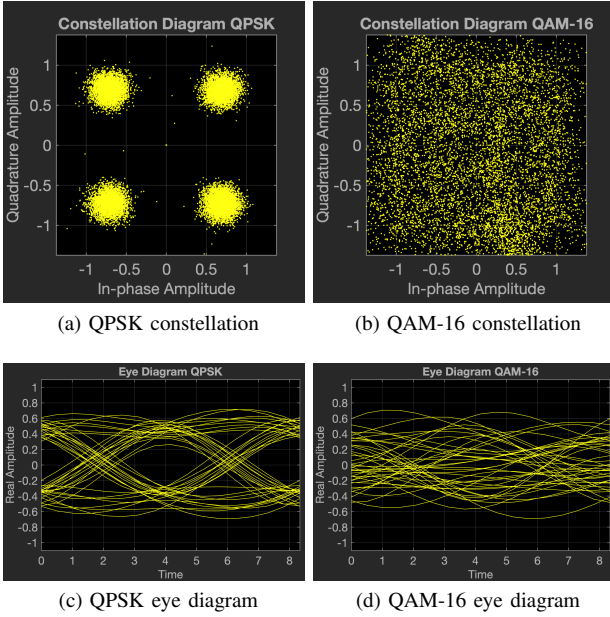


Fig. 10. Eye diagram and constellation for QPSK (a and c) and QAM-16 (b and d) modulated symbols. Transmit power:  $-25\text{dBm}$

TABLE V  
MEASURED PERFORMANCE PARAMETERS. TRANSMIT POWER:  $-10\text{dBm}$

System Parameter	Measured Value
Modulation	QPSK / QAM-16
EVM	$-17.5\text{dB} / -14.5\text{dB}$
BER	$1.35 \cdot 10^{-5} / 6.0 \cdot 10^{-2}$

TABLE VI  
MEASURED PERFORMANCE PARAMETERS. TRANSMIT POWER:  $-25\text{dBm}$

System Parameter	Measured Value
Modulation	QPSK / QAM-16
EVM	$-15.9\text{dB} / -12.8\text{dB}$
BER	$1.4 \cdot 10^{-4} / 1.32 \cdot 10^{-1}$

### C. Discussion of Obtained Results

The proposed system is supposed to broadcast speech data with adaptable sound quality, by switching between QPSK and QAM-16 modulation based on the detected error rates. During the full system test, the threshold for changing state was set to  $1 \cdot 10^{-2}$ . With this threshold, the system changed quality state when the transmit power was reduced to about  $-17\text{dBm}$ . The test verified that the adaptable quality feature works as expected. The measured 5ms delay show that the system is well suited for two-way communication as well as broadcasting.

When transmitting QPSK modulated data at low data rate, the system delivered almost noise free sound, at both power levels. The sound quality was not audibly reduced by the transmission. Figure 9 and 10 and table V and VI supports this qualitative description.

The transmission of QAM-16 modulated data was not equally successful. At the highest power level, the sound was clearly audible, and spoken words recognizable, but the sound was covered in noise. At the lowest power level the sound was almost completely lost in noise. This result is again supported by figure 9 and 10 and table V and VI. The constellation diagram in figure 9 (b) shows a higher noise level for QAM-16 than for QPSK, which is also reflected by the EVM. The constellation also shows that the accuracy is worse for the symbols with strongest amplitude. The constellation is also slightly rotated, showing that the phase synchronization algorithm was not as accurate on the QAM-16 symbols as for QPSK.

One major advantage of QPSK modulation is the constant symbol amplitude, which reduces nonlinear effects in the amplifiers of the USRP. For QAM-16, the amplitude is not constant, which is giving rise to nonlinear effects in the RF Hardware. The nonlinear effects tend to increase with amplitude, which is in line with the constellation of figure 9 (b). We therefore suspect that these nonlinear effects is a major reason for the poor system performance with QAM-16 modulation. Figure 9 (b) also show that the corner symbols seem to be suffering the most from these nonlinearities. As the training sequence used for phase synchronization is transmitted at these corners only, this may explain why the phase is inaccurately estimated.

For further verification of the system, more reliable hardware should be considered in order to reduce the nonlinear effects. To make the phase estimation more robust one could also reduce the amplitude of the training symbols.

### VII. CONCLUSION

The design and implementation of a radio communication system for broadcasting of speech with adaptable data rate is presented. The system adapts the transmitted data rate to the state of the radio channel by evaluating detected bit error at the receiver. The transmitted data rate is varied by a factor 2 by switching modulation format between QPSK and QAM-16.

The system was verified in an indoor environment with a transmission distance of 1 meter and variations in the radio channel is simulated by adjusting the transmit power. The system transmit at high data rate (520, 8 kb/s) when using QAM-16 modulated symbols at  $-10\text{dBm}$  and low data rate (249 kb/s) for QPSK at  $-25\text{dBm}$ . Bit error rates of  $6.0 \cdot 10^{-2}$  and  $1.4 \cdot 10^{-4}$  was measured at high and low data rate respectively.

The high BER for QAM-16 modulation makes the effect of adaptable quality hardly audible, because of the high noise level when transmitting at high data rate. The adaptable quality feature is however interesting in itself and this demonstration shows a working “proof of concept”. By adapting the transmitted data rate to the state of the radio channel, the system yields a better utilization of available resources. This two-level adaption could be extended to several levels for higher performance in future improvements.

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APPENDIX A  
BLOCK DIAGRAM

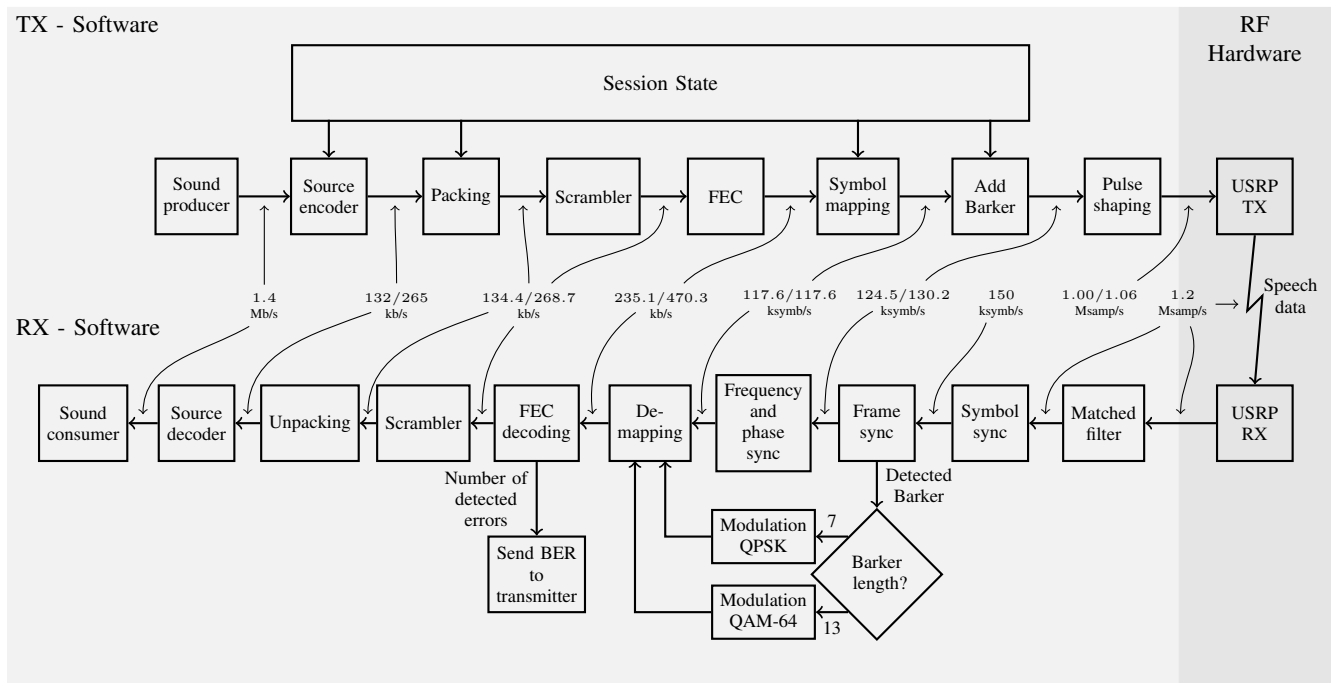


Fig. 11. Block diagram of data packet system. This block diagram shows the forward path of the system, where speech data is being transmitted. Data rates for Low/High data quality is indicated on each arrow.

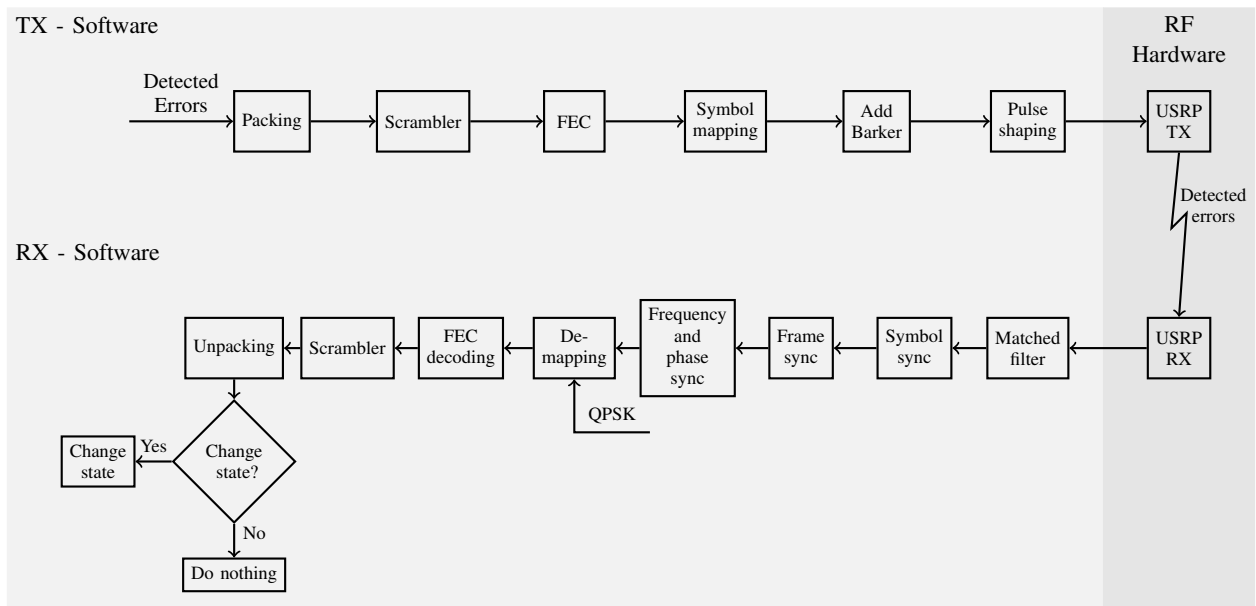


Fig. 12. Block diagram of BER packet system. This sub-system constitutes the feedback path where the receiver transmit information about detected error rate back to the transmitter.