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 $\frac{E_b}{N_0}$, this would however 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 obtained BER back to the transmitter. This enables system to adapt the data rate to 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 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. Specifications

Some key specifications is listed in table I. The values are listed for low / high data rate transmission. 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.

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. Block diagrams for these subsystems is shown in appendix C and the behaviour will be explained in the following subsection.

TABLE I
SYSTEM SPECIFICATIONS

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 / 264,6 kbits/s
Channel coding	Hamming (4,7)
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

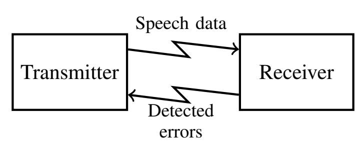

Fig. 1. Top level block diagram of proposed system. Speech data is sent in the forward path from transmitter to receiver and the number of detected errors is sent back from receiver to transmitter.

The burst format is shown in figure **??**.

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.

Training sequence	Header	Speech data	Guard bits
52	11	256	12

(a)

Training sequence	Header	Speech data	Guard bits
156	11	768	36

(b)

Fig. 2. Burst format for QPSK (a) and QAM-64 (b) modulation

B. Source Encoder

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 **??**). In addition the sample rate are reduced by decimation with a factor of 2 or 4 for high and low quality respectively.

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 very fast, pre-written C-code.

E. Symbol Mapping

The system uses Grey Code for symbol mapping

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 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.

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 [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, 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 [4]. 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 V.

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 1 and 2 respectively.

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

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

IV. Design Motivation

As stated in section **??**, the purpose of the proposed system is to demonstrate a radio system with adaptable data rate and this is the background for all design choices that is made.

V. 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] 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).
[3] Siretta, “Delta-7a datasheet,” https://www.siretta.com/wp-content/uploads/2020/01/Delta-7A-Rev-1.7.pdf, (Accessed on 04/14/2020).
[4] 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
System specifications

Appendix B
Link Budget

Appendix C
Block Diagram

Appendix D
Symbol Mapping Scheme

TABLE II
SYSTEM SPECIFICATIONS

System Variables	Variable	Units	Equation	Value	
				Low Data Rate	High Data Rate
Frequency	f_0	MHz		2415	2415
Modulation				QPSK	QAM64
Bit per symbol	m			2	6
Sound sampling rate	f_s	Hz		11025	22050
Bits per sound sample	b_s	bits		8	12
Sound datarate	R_{ss}	kbits/s	$f_s \cdot b_s$	88,2	264,6
Channel coding				Hamming (4,7)	Hamming (4,7)
Channel coded data rate		bits	$R_s s \cdot 7/4$	154,35	463,05
Packet Parameters					
Packet header size		bits		11	11
Packet data length		symbols		128	128
Packet size		bits		256	256
Frame Parameters					
Training sequence type				Barker	Barker
Training sequence length		symbols		26	26
Training sequence size bits		bits		319	935
Frame size		bits		319	935
Burst Parameters					
Guard period		symbols		6	6
Burst size		bits		331	935
Transmission Characteristics					
System bit rate	R_b	kbits/s		187.39	572.67
Symbol rate	R_s	ksymbols/s		93.69	95.45
Pulse shaping filter				root raised cosine	root raised cosine
Pulse shaping filter parameter	α			0,3	0,3
Minimum signal bandwidth	Δf	kHz		**60.9**	**62.0**

TABLE III
LINK BUDGET

TX Loss	Variable	Units	Equation	Value	
				QPSK	QAM64
PA Power	P_{PA}	dBm		10	
TX Connector Loss	L_{ConT}	dB	from connector data sheet	-0.3	
TX Power	P_T	dBm	$P_T = P_{PA} \cdot 2L_{ConT}$	9.4	
TX Antenna Gain	G_T	dBi	from antenna data sheet	3	
Effective (Isotropic) Radiated Power	EIRP	dBm	EIRP $= P_T G_T$	12.4	
Path Loss, ITU Indoor Propagation Loss Model					
Distance	d	m		10	
Floor loss factor	$Pf(n)$	dB		0	
Distance power loss coefficient	N			38	
Total ITU path loss	L_P	dB		-77.66	
RX Loss					
RX antenna gain	G_R	dBi		3	
RX connector loss	L_{ConR}	dB		-0.3	
Total RX Loss	L_R	dB		2.4	
Total Received Power	P_R	dBm	$P_R =$ EIRP $\cdot L_P \cdot L_R$	-62.86	
RX Noise					
Antenna Noise Density	N_0	dbm/Hz	Measured with spectrum analyser. Average of several single runs	-145.73	
Antenna Total Noise Power	N	dBm	N0 $\cdot$ BW	-97.806	
RX Noise Figure	NF	dB	From NI USRP-2901 datasheet	7.000	
Small Scale fading margin	M_{ssf}	dB	From measurements of RX power variations	9.400	
RX Properties					
Carrier-to-noise ratio	C/N	dB	$C/N_0 = \frac{P_R}{NNFM_{ssf}}$	18.548	
Eb over N0	E_b/N_0	dB	$\frac{E_b}{N_0} = \frac{C}{N}\frac{\Delta f}{R_b}$	13.666	8.893
Eb over N0	E_b/N_0	lin		23.262	7.749
Bit error rate	BER			8.56E-08	3.17E-04

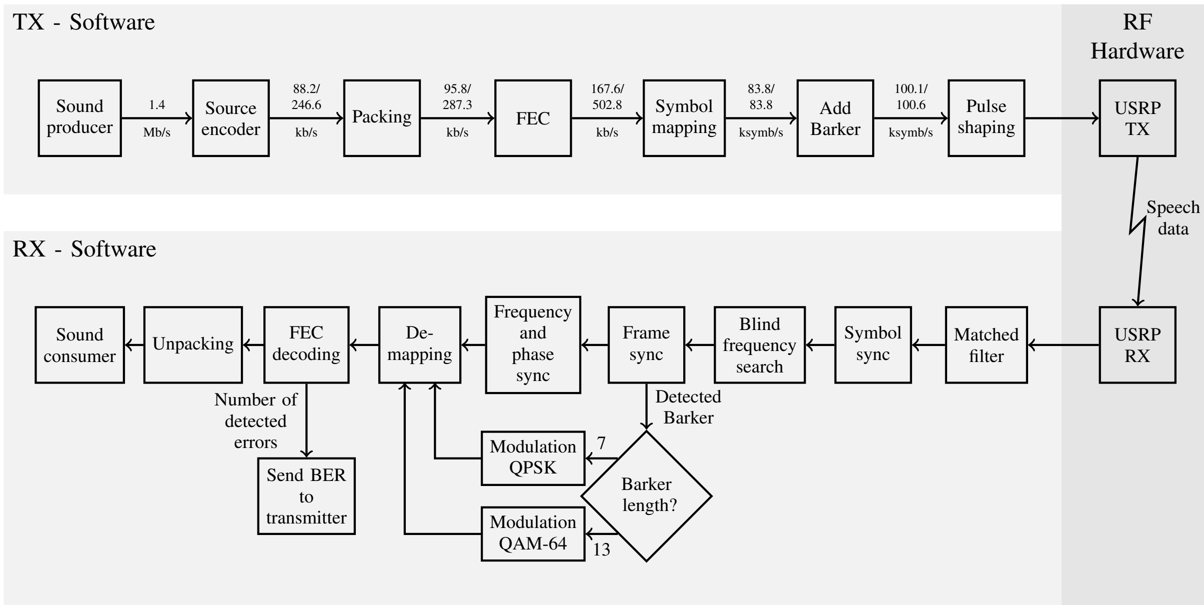

Fig. 3. Block diagram of data packet system. This block diagram shows the forward path of the system, where speech data is being transmitted.

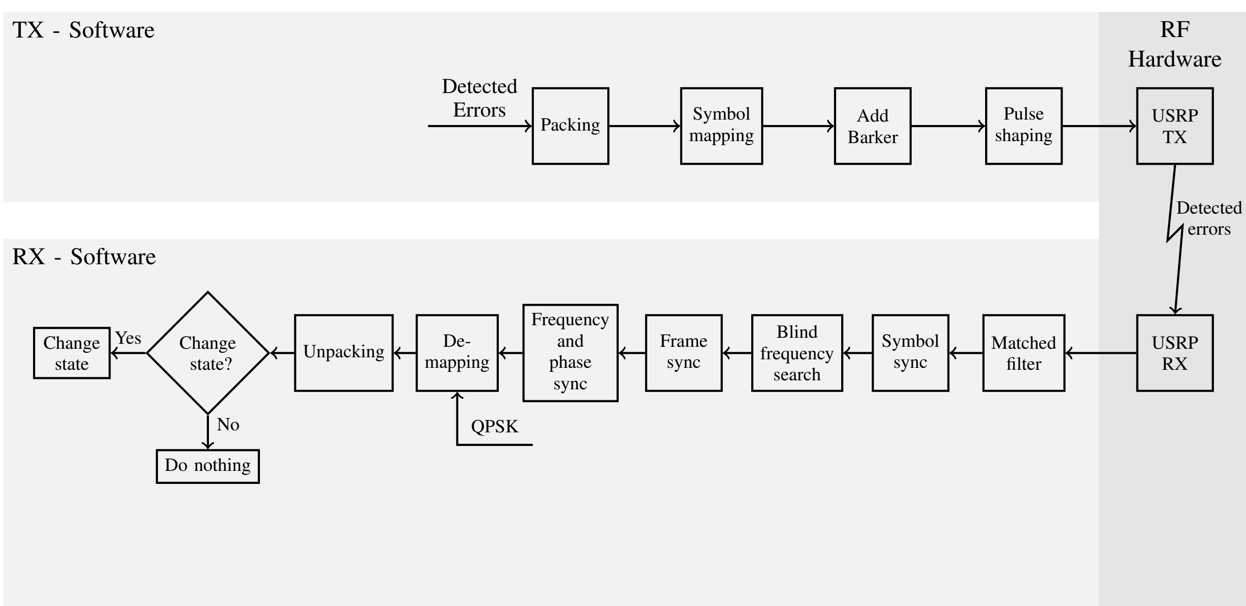

Fig. 4. Block diagram of BER packet system. This sub-system constitutes the feedback path where the receiver transmit information about detected error rate back to the transmitter.