# Repeater Data Transmission System

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## Synopsis

The paper describes a specialised data communications system. The data system has the capability of carrying low data rate information over an analogue radio channel at the same time as voice traffic. It is based on the capstone project I completed in order to receive my Bachelor of Engineering Degree[1].

### Summary

For many years Radio Amateurs have been developing voice repeaters and their associated links and controllers. Gradually these systems are becoming more and more sophisticated. Unfortunately, much of the modern digital technology is financially beyond the reach of Amateur Radio clubs.

Albeit the technology being developed in this project is not a ground breaking fully digital solution, it provides a bit of the flexibility of digital control over the existing analogue radio links. This is provided at a cost that is affordable and realistic for amateur radio clubs who want to go the step beyond simple linked repeaters.

This system in essence is a FDM (Frequency Division Multiplexed) system, where by the base-band frequency range of 20 Hz to 4.5 kHz is being utilised to carry the voice band modulation. The 5.5-6.5 kHz frequency range is being used to carry 200-300 bit per second data.

This base-band modulation is compatible with many existing narrow band frequency modulated (NBFM) radio links.

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[1] The full documentation (with circuit diagrams) is available from www.tapr.org

The purpose of this data link is to allow communication between intelligent microprocessor based repeater controllers at each linked site. This digital communication will carry messages about system mode changes, signal strength information for receiver voting and other telemetry and logging data.

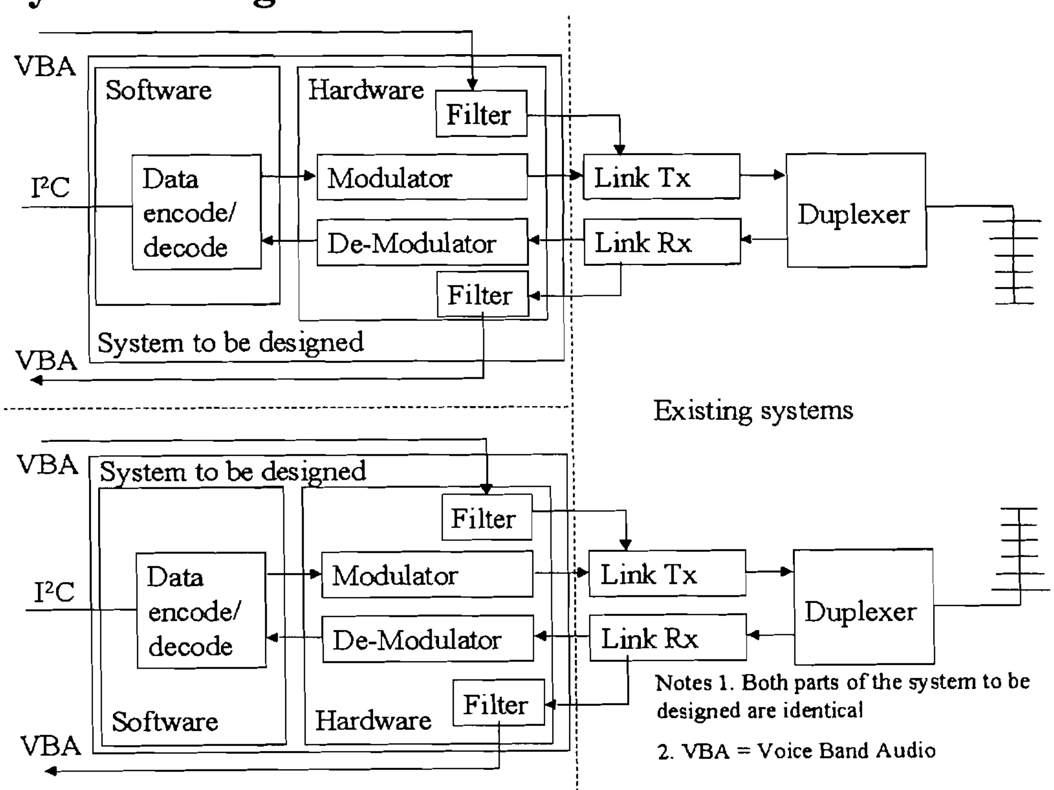
## Introduction

This project has a few overall specifications that should be outlined first. The aim is that the outcomes of this project, when appropriately published, will be useable to all Amateur Radio Repeater groups and individuals. With this firmly in mind it must be considered for a moment, the technical ability and purchasing ability of a majority of repeater minded amateurs.

Not many active amateurs are trained specifically in Electrical Engineering by university or TAFE. This immediately suggests that any designs should be sufficiently simple in overall complexity that amateurs in general can build and maintain them.

Most amateurs do not have the range of parts suppliers available to them that a practising Electrical Engineer does. Consequently all parts used where possible should be restricted to those that can be purchased where possible from domestic suppliers, such as Dick Smith Electronics, Jaycar and Altronics.

## System Design



The full duplex analogue radio links that the signalling system is conveyed over are in many cases pre-existing infrastructure. Currently these links operate in a simple carrier operated mode. This system will pass data over the links simultaneously with the voice traffic. The system provides an $I^2C$ interface to communicate with the other controllers and an audio interface for connection to the radio communications transmitters and receivers.

The prototypes implement 300 bit per second data rates as this allows testing and debugging of the software using standard serial interfaces on a PC.

## Data Types

There are three different types of data to be transmitted over the link. Each type of data requires different handling. The packet types are

1. Time and error sensitive
   These packets will usually be the change in state of a remote receiver's carrier detector. This data needs to be passed quickly (within 100ms) without error, so any packet containing errors will need to be resent.

2. Time sensitive
   These packets will mainly be a value indicating the quality of a remotely received signal that is to be voted. It is essential that these types of packets are passed in a timely manner. They will tend to emanate at about a 20Hz rate when there is voting occurring. If an error occurs the packet in error is to discarded.

3. Error sensitive
   These types of packets are usually mode changes that are user initiated or telemetry data for logging purposes. This type of packet will be divided up into two sub types of packet

   a. The mode changes that need to be transferred within a few seconds (realtime)
   b. Telemetry data for logging that will be transferred only if there is space data capacity or the data channel is idle.

   If errors occur on either of these types of packets, they need to be corrected by a process of retries.

## Error control

Error control is required for type 1 and 3 packets. The error detection methods is able to detect at a minimum of one bit error in a packet. This specification has been relaxed for the initial prototype of the software to ensure it is operational in time.

### Packet queuing and Retries

The above 3 types of packets need to be queued and transmitted with a priority that is appropriate to their urgency. Packet type 1 has the highest priority, and packet type 3 has the lowest priority.

The software receives commands from the other interfaced microcontrollers and queue them in appropriate order for transmission. Incoming commands are separated into 3 buffers for processing. Retries need to be used if type 1 and 3 packets fail with an error in the data received. The retry is given the same priority as the packet that it is trying to recover. Only type 1 and 3 packets need acknowledgment.

## Audio spectrum use

All radios to be used with this system have their audio levels both receive and transmit adjusted such that 0dBm on the baseband inputs and outputs corresponds to peak deviation (in most cases 5kHz deviation).

### Spectral content

Voice band audio covers a range of 20Hz to 4.5kHz +0 –3dB Signalling data uses a maximum bandwidth of 1kHz centred on 6kHz. The signalling carrier should not exceed –10dB with respect to the radio systems peak deviation.

### Signalling content in voice band audio

All signalling artefacts are greater than 40dB below peak deviation in the demodulated and filtered audio that is presented to the repeater from the link. All voice band audio are greater than 13dB below the signalling carrier level in the signalling bandpass.

### Signalling data content in voice band audio post demodulation

Once the Signalling data is recovered from the baseband audio from a link receiver, the signalling data needs to be removed to leave the voice band audio. The maximum level of signalling data to be present in a voice band audio line is at –40dBm.

## Hardware

### Modulator

Inputs:

- Data stream: 5 V logic

Outputs:

- Signalling data: audio line capable of driving 600 ohm load level to be determined by simulation, suggested range from –30dBm to –10dBm.

Functionality:

The modulator produces an amplitude modulated carrier centred on 6kHz with an overall bandwidth of no greater than 1kHz. The modulator is to operate at a minimum symbol rate of 200 symbols per second with two amplitude levels.

### Demodulator

Input:

- Signalling data: audio line terminating in 600 ohms. Sensitivity of signalling decoder between –30dBm and –10dBm to be determined by simulation. Input signal will contain voice band audio up to 0dBm in level.

Outputs:

- Data stream: 5V logic same sense as input to modulator

Functionality:
The demodulating should be capable of fully recovering the data encoded by the modulator.

### Voice band audio filtering and radio interface

The voice band audio filtering is applied at two points in each path. It is applied to the voice band audio path before the signalling data is inserted, and when the signalling data has been recovered from the baseband audio.

This filter needs to pass the audio range 20Hz to 4.5kHz +0 -3dB. There must be greater than 30dB of attenuation across the signalling data band pass of 5.5-6.5kHz. Above 6.5kHz the filter response may rise as high as a loss of 6dB and then it falls off at a minimum of a first order rate.

### Protocol

A cost-effective microprocessor from Atmel was chosen for this part of the system. The system will have to communicate with other microprocessors over a bus based communications system. This microprocessor has the task of looking after all the signalling data coding, error correction, and other signalling data house keeping issues.

## System Design and Implementation

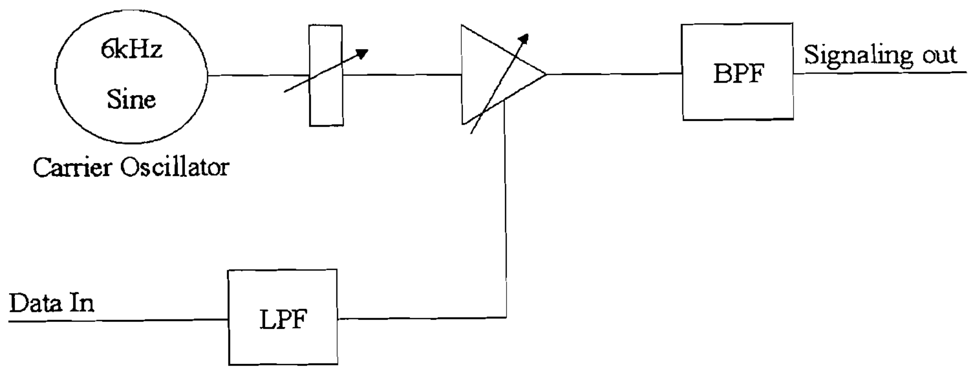
The full duplex analogue radio links that the signalling system will be conveyed over are in many cases pre-existing infrastructure. Currently these links operate in a simple carrier operated mode. This system passes data over the links simultaneously with the voice traffic. The system provides and $I^2C$ interface to communicate with the other controllers and an audio interface for connection to the radio communications transmitters and receivers.

As the specification has stated a data rate of at least 200 bits per second needs to be achieved. The prototypes will implement 300 bit per second data rates as this allows testing and debugging of the software using standard serial interfaces on a PC.

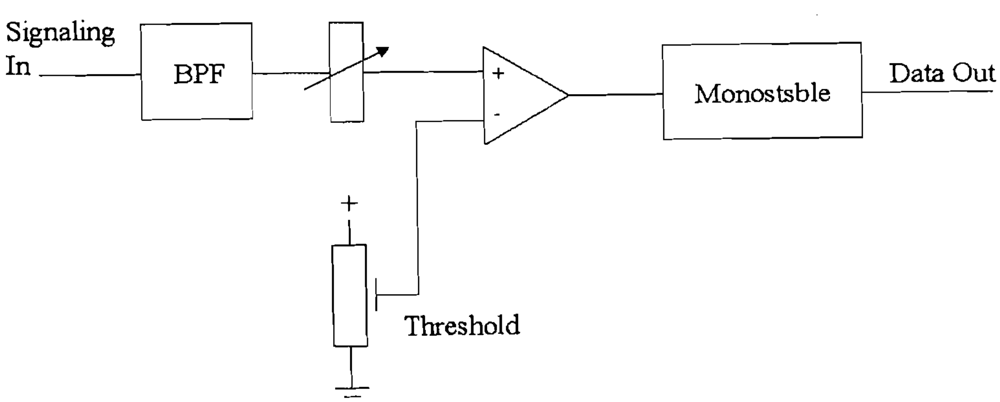
## Hardware

The proof of concept prototype provided some confidence in the ability of the radio links to handle the above audible audio signalling. It also provided the basis for the block diagram of the system.

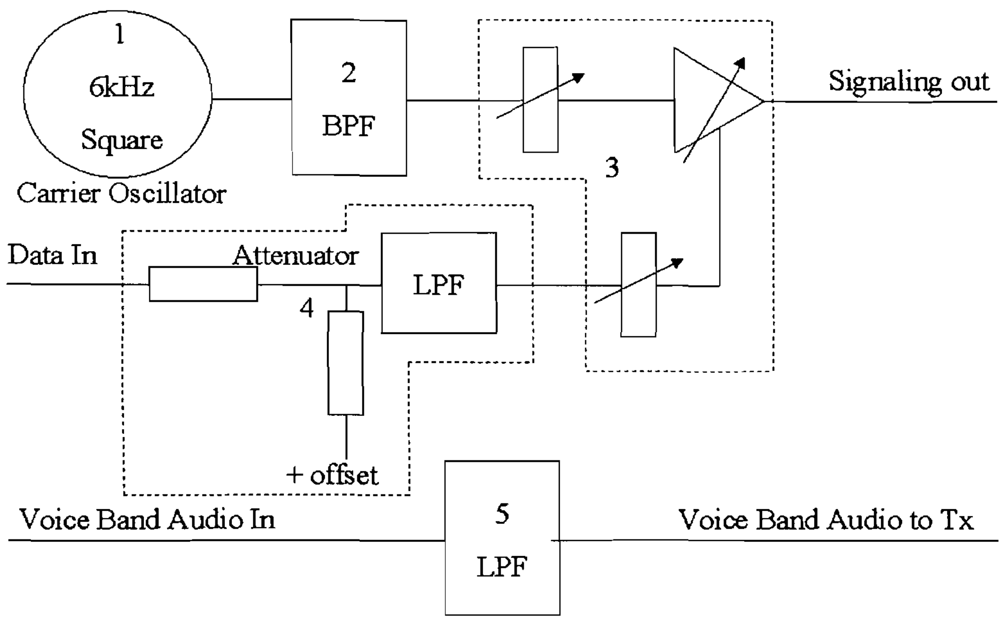
Transmit Modulator - DSB with carrier



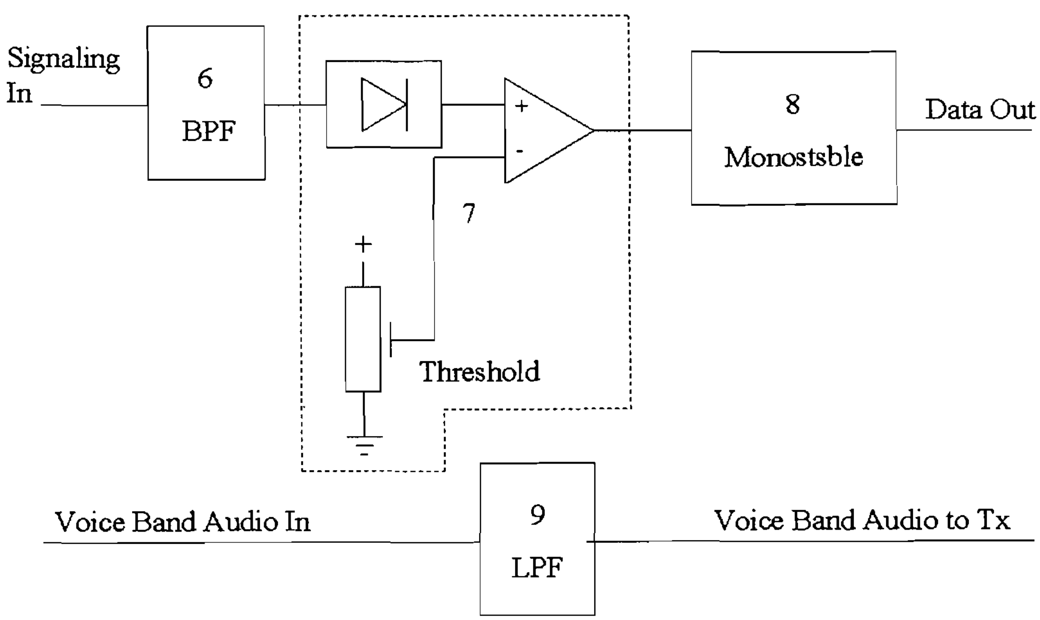

Receive Demodulator - DSB with carrier

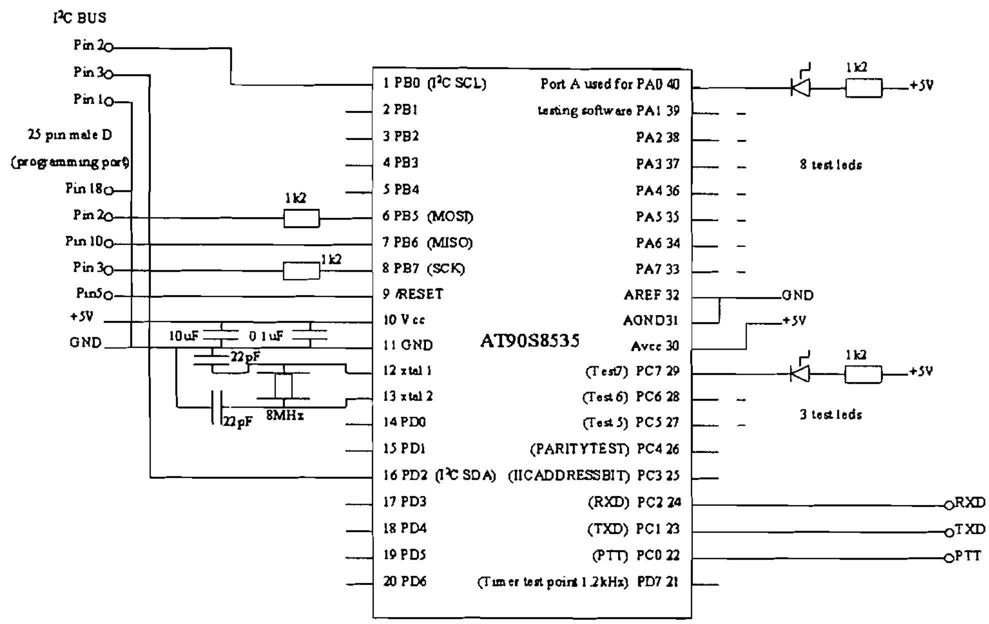


Final Transmit Modulator - DSB with carrier




Final Receive Demodulator - DSB with carrier






# Design Realisation

For the implementation phase of the process the actual circuit topologies are determined. As a number of the blocks are audio frequency filters, the topology's being used are pretty well defined and tried and true. The Active filter cookbook was used as a reference on the topic. This provided the normalised value sets to allow for easy scaling of the filters to the desired operating frequency.

The frequency scaling was largely based on fixed capacitor value and obtaining the precision of tuned frequency with the resistor. This was the case as the local hobbyist electronics suppliers have a single value of precision capacitor value available. Dick Smith Electronics stocks a 1nF 1% polystyrene capacitor. Due to a number of the filters being of order 4 or higher, this capacitor was chosen as a necessity to ensure reproducibility. Resistors are easily procurable with a 1% tolerance in the E24 series so these were varied to obtain desired frequency and damping characteristics.

The next step of circuit simulation only applied to the AC based circuits such as the active filters, as the simulator available only support AC analysis. The simulator being used is the ARRL Radio Designer. This simulator, as the name suggests, is aimed at Radio Frequency based circuit analysis, hence the specialisation on AC analysis. This package takes a simple net list entry and allows production of a whole variety of plots. The major plots of interest for this project are the bode-plots and monte-carlo analysis.

The monte-carlo sensitivity analysis reveals whether the filter will meet the specification in a reproducible manner. At this point if non-compliance to the specification is found the design is adjusted and filter simulation repeated

Following the completion of the design stages, the circuit is built to confirm operation. The prototypes were constructed on vero-board. This step reveals if there are any issues not identified in the design and simulation stages.

### Result of testing

On the whole the signal processing hardware meets specification. There is one non compliance with specification in the form of the loss at 4.5kHz in the voice band audio filter. This miss of specification was accepted from the design and simulation stage as being an acceptable deviation from the specification. This deviation was due to the additional complexity in meeting specification, compared with one of the projects objectives being to minimise complexity and cost. Despite this the hardware in all sections agreed with the simulated results.

This is an experimental design project, and the final result is a simplified version of the final desired outcome for the longer term. The final outcome involves work levels that exceeded the scope and requirement for this project, hence the specifications were relaxed to allow the initial functional system to be created within the time frame of this project.

# Conclusion

This project provided a good mix of work that was hard, challenging and problematic for me as well as some straightforward parts. The hardware components of the project, provided for a well executed and on time section of the project. The hardware area is where I have had most of my experience in the past, so this part of the project came to me pretty easily.

The software sections provided a vast number of problems, and completely destroyed any chance of achieving the desired time frames and time line of this part of the project. I also learnt the most from this section of the project. The number of hidden problems that came to the surface meant that the software work was definitely a very good learning exercise.

As this project was always an experimental design that is not producing a finished product within the scope of the capstone project, there is room for further work beyond the scope of the capstone project. Some of this work includes, further iteration and fine tuning of the voice band audio filter design. There is also room for expanding the behaviour of the software to be a little more intelligent with regard to the traffic being handled and the error correction systems. For example, this project implemented simple single bit parity error checking.