Experimental data transmission method using sound

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1 Introduction

A method for transmitting and receiving 20 bytes long data packets using sound is described. The method uses continuous-phase multiple frequency-shift keying (CPFSK), and a robust synchronization method. A Reed-Solomon code is used for forward error correction.

A software was written in the Python language for experimental and demonstration purposes. Scripts are provided to send and receive short text messages using the computer's audio devices, or to read/write the audio to/from data files. The files can be played back, or converted into more traditional audio formats using sox[1].

A framework was established to gauge the performance of the software. A script can generate random messages and save them into audio files with various levels of white noise. An other script then can be used to attempt decoding these files. Various statistics are shown after the script finished decoding the files.

Experiments show that this method can transmit data 99.9% accurately at a carrier to noise ratio (CNR) of -11dB (or at about 10dB E_b/N_0) measured over 22.05 KHz bandwidth, and with 83.4% accuracy at -12dB CNR (about 9dB E_b/N_0).

In real-world experiments using a small set of consumer-grade computer audio devices reliable data transfer could be achieved over 1 meter in a quiet room at quite low volume levels. Noisy environments mandate higher volume setting and/or shorter distances. Interference due to multipath propagation probably is the main performance bottleneck.

2 Theory of Operation

I provide a summary on the operation of the software. See the in-line code documentation for implementation details.

2.1 Modulation

The data has to be transmitted and received with equipments designed to produce and record audio. Since little was known about the working environment, an Additive Gaussian White Noise (AGWN) channel was used as the channel model. In real life however, it is likely that noise bursts will occur frequently.

Audio sampling rate was determined to be 44.1KHz. This sampling rate along with 48KHz is the most frequently supported one, and 48KHz audio can easily be downsampled on devices only supporting that.

The channel properties were chosen so that the transmitting device could be a low-quality piezoelectric speaker. A carrier frequency of about 4KHz, and a maximum bandwidth of also 4KHz was determined. The bandwidth were kept lower than the carrier frequency, to prevent harmonics interfering with the reception, so the transmitter can drive the speaker with a square wave.

Using 44100 samples per second sampling rate, a 200 sample DFT will have a bucket size of 220.5 Hz. To keep the modulation process simple, one transmitted symbol will contain 4 bits, or half a byte of information. To encode 4 bits, a symbol must communicate 16 separate states.

The modulation uses 17 frequencies in the form of $4410 + n * 220.5$ Hz where $n \in [0..16] \cap \mathbb{Z}$. About half of the time the carrier ($n = 0$) is transmitted, and no actual data is sent. This is for synchronization. When this tone is off, one of the other 16 frequencies ($n \in [1..16] \cup \mathbb{Z}$) is transmitted, this is when data is actually being transmitted. So instead of 16 symbols, a 17 symbol alphabet is being used, out of which the carrier or zero symbol is used for frame synchronization.

The carrier is turned on-off in a way so it can easily be found using correlation. The exact sequence of on-off states affect how easily it may be spotted by correlating a potentially noisy and arbitrary time-shifted signal with a known, clean synchronization sequence.

Good sync sequences were found using gradient descent with random restarts: essentially by checking random sequences, and looking for neighbors with increasingly better properties, until the process stuck in a local optimum, after which it is restarted from an other random vector.

A total of 200 symbols are transmitted, out of those 108 are used to transmit data, and 92 used for synchronization only. Transmitting one frame takes about 0.907 seconds. **The effective bit rate is 176.4bps.**

2.2 Demodulation

Demodulation begins by recording a chunk of audio longer than a single frame. After normalizing the received data and correlating it with the synchronization sequence, a decision is made whether there is a signal present. The correlation will return the position where the sync signal matches the audio the most.

Using the position information 108 DFTs at positions where the first tone is off. The received power in each bin is observed, and a hard decision is made about what symbol the particular 200-sample audio data might encode. Only the DFTs encoding the 108 data nibbles are kept.

2.3 Forward Error Correction

A Reed-Solomon code RS(54,20) were chosen for forward error correction, mainly because implementations were readily available, and of course because of their good performance.

The error correcting code takes 20 byte payload, and produces a 54 byte codeword. The code is capable of correcting 17 errors or 34 erasures. Currently the software does not signal erasure towards the decoder, but there are methods[3] for doing so.

3 The Software

This section describes how the software should be operated to run demonstrations and experiments.

3.1 Prerequisites

The demonstration software requires a functional Python 3.6 installation, and a machine with enough resources. The following configuration *before* Python installation is recommended:

- Intel Core i5 processor, or better
- 4GB RAM
- 6GB of free disk space
- Speakers
- Microphone
- Ubuntu Linux 17.04 or newer is recommended, a recent Windows or macOS should also work

It is recommended to install a Python distribution called Anaconda [2]. Anaconda a contains stable and fast Python build along with several important libraries.

3.2 Installation

Anaconda 5.0.1 with Python 3.6 can be downloaded from:

`https://www.anaconda.com/download`.

There are installers for Linux, Windows and macOS operating system. Any of those systems should be able to run the demo software, but it had only been tested throughly on Ubuntu Linux ”zesty” 17.04.

The scripts `send.py` and `receive.py` use *pyaudio*, a Python binding for an audio library called *portaudio*. The portaudio library (.so, .dll, or .dynlib) and header files need to be installed. The installation process differs across operating systems, and can be as simple as issuing a command (Linux and maxOS), or having to install Visual Studio and compile the library by hand (Windows)[1].

3.2.1 Installation on Ubuntu Linux ”zesty” 17.04

The `portaudio19-dev` package need to be installed with the package manager `apt`:

```
$ sudo apt install portaudio19-dev
$ cd <path-to-demo-software>
$ pip install -r requirements.txt
```

3.2.2 Installation on Apple macOS

Portaudio need to be installed using `brew`:

```
$ brew install portaudio
$ cd <path-to-demo-software>
$ pip install -r requirements.txt
```

3.2.3 Installation on Microsoft Windows

On Windows, the portaudio binaries are included in the pyaudio wheel, so separate installation should not be necessary (make sure `pip` and `python` are on the `PATH`):

```
C:\> cd <path-to-demo-software>
C:\> pip install -r requirements.txt
```

[1] This is why the .dll-s and headers bundled with the pyaudio Python library for Windows

3.3 Usage

The scripts `test-gen.py` and `text-score.py` can generate a large set of audio files, and run a decoding benchmark on them respectively. Any computer capable of running Python properly may be used to run the benchmark. Since the process involves generating 14000 files consuming about 1.2GB space, this should be done on a fast machine.

The scripts `send.py` and `receive.py` use audio devices to play and record audio. They use the default system device, but they can set up to use any device with the `-d <number>` or `--device <number>` parameter. Use `devices.py` to view a list of audio devices.

All command examples assume Linux operating system, but they should work with no or only minor modifications on macOS and Windows too. Commands should be executed in the directory where the scripts are located.

3.3.1 Listing audio devices

The `devices.py` script shows a list of audio devices available. On some systems error messages may occur on the first few lines. These can be ignored.

Since `send.py` and `receive.py` are using the default audio device, this list will probably rarely be needed.

```
$ ./devices.py
ALSA lib pcm.c:2495:(snd_pcm_open_noupdate) Unknown PCM cards.pcm.rear
ALSA lib pcm.c:2495:(snd_pcm_open_noupdate) Unknown PCM
cards.pcm.center_lfe
ALSA lib pcm.c:2495:(snd_pcm_open_noupdate) Unknown PCM cards.pcm.side
ALSA lib pcm_route.c:867:(find_matching_chmap) Found no matching
channel map
0 HDA Intel PCH: ALC898 Analog (hw:0,0)
1 HDA Intel PCH: ALC898 Digital (hw:0,1)
2 HDA Intel PCH: ALC898 Alt Analog (hw:0,2)
3 USB Audio CODEC: - (hw:1,0)
4 HDA NVidia: HDMI 0 (hw:2,3)
5 HDA NVidia: HDMI 1 (hw:2,7)
6 HDA NVidia: HDMI 2 (hw:2,8)
7 HDA NVidia: HDMI 3 (hw:2,9)
8 sysdefault
9 front
10 surround21
11 surround40
12 surround41
13 surround50
14 surround51
15 surround71
16 iec958
17 spdif
18 pulse
19 dmix
20 default
$
```

3.3.2 Sending or saving messages

The `send.py` script can send a message by playing the modulated audio carrier on an audio device, or saving and audio file.

In it's simplest form it's just called with a short text to send:

```
$ ./send.py test
CNR (dB):     inf
Eb/N0 (dB):   inf
Peak:         0.252437119188
$
```

The command always displays the estimated carrier to noise power ratio and E_b/N_0 in decibels. Since there is no noise added by default, both figures are displayed as being infinite.

Adding noise, distortion and saving the audio into a file:

```
$ ./send.py -f test.s16 -c -11 -d 10000 test
CNR (dB):     -11.0397105534
Eb/N0 (dB):   9.92938957672
Peak:         0.742417642773
$
```

The files store a single audio channel at a sampling rate of 44100 KHz. Each sample is a signed 16 bit little endian integer. The files contain no header or any kind of metadata.

One can specify a string of hexadecimal digits to send with the `-x` or `--hex` flag:

```
$ ./send.py -f test.s16 -c -11 -d 10000 -x \
  9d7133b7f07274a5be88a06f7b5ae2914822c008

CNR (dB):     -11.0397105534
Eb/N0 (dB):   9.92938957672
Peak:         0.742417642773
$
```

For a brief description of usage and parameters, issue `./send.py -h`.

3.3.3 Receiving or loading messages

The script `receive.py` can be used to load audio files, or decode audio read from an audio device in real-time. The test.s16 file created above can (very likely) be decoded with the following command:

```
$ ./receive.py -f test.s16 -x
Got sync signal, decoding...
DECODED: >>> 9d7133b7f07274a5be88a06f7b5ae2914822c008 <<<
End of data stream reached, terminating
$
```

Without the `-f` parameter the script listens on the default audio device. The `-x` flag is useful for printing arbitrary binary data. Without this flag the script tries to convert the received data to ASCII strings, and print that instead of hex digits. Call the script with the `-h` parameter to get a list of usable flags and parameters and their descriptions.

3.3.4 Recording audio

A short script is provided for Linux using the SoX command `rec`, and providing it with the necessary parameters, so only the file name has to be given:

```
$ ./rec.sh test.s16

Input File     : 'default' (alsa)
Channels       : 1
Sample Rate    : 44100
Precision      : 16-bit
Sample Encoding: 16-bit Signed Integer PCM

In:0.00% 00:00:05.20 [00:00:00.00] Out:221k  [ -====|====- ]     Clip:0
Aborted.
$
```

The recording can be stopped by hitting CTRL+C. It is worth keeping an eye on the ”Clip” indicator at the end of the last line. If this counter starts growing then the audio level is too high.

3.3.5 Playing saved audio

Playing audio is also handled by SoX via a Linux shell script:

```
$ ./play.sh test.s16

test.s16:

 File Size: 459k      Bit Rate: 706k
  Encoding: Signed PCM
  Channels: 1 @ 16-bit
Samplerate: 44100Hz
Replaygain: off
  Duration: 00:00:05.20

In:100%  00:00:05.20 [00:00:00.00] Out:229k  [      |      ]     Clip:0
Done.
$
```

3.3.6 Generating test files

The test set generator script will generate one thousand random test files for each CNR level. The tested levels are between and including -14dB and -1 dB.

The `test-gen.py` script accepts no arguments, and calling it will start generating test files right away:

```
$ ./test-gen.py
  0%|                              | 5/14000 [00:01<1:06:18,  3.52it/s]
$
```

During test set generation a progress bar with the speed and ETA is displayed. Before generating a test set, make sure the `data` directory only contains a `.gitignore` file, and no *.s16 audio files. These are not removed automatically, and new files will have different names every time.

Every file is exactly 1 second, or 44100 samples – 88200 bytes – long. The signal is right in the middle of every file at offset 2050.

3.3.7 Scoring performance on a test file set

The `test-score.py` script will read any .s16 files from the data directory and will attempt to demodulate and decode them. It will record any success or failure in finding the synchronization signal or decoding the actual message.

Before exiting, the script will print out a rather dense representation of the results. It looks like this:

```
FILES:          {-5: 1000, -3: 1000, -4: 1000, -14: 1000, ...
GOOD DECODES:   {-5: 1000, -3: 1000, -4: 1000, -12: 834, ...
NO DECODES:     {-14: 1000, -13: 897, -12: 166, -11: 1}
BAD DECODES:    {}
GOOD SYNCS:     {-5: 1000, -3: 1000, -4: 1000, -14: 980, ...
BAD SYNCS:      {-13: 10, -14: 20, -12: 1, -11: 1}
BAD SYNC SET:   {2040, 2060}
```

Each line shows a metric. The number before the colons are specific CNR values, the number after the colons are the number of occurrences.

For example the „FILES" line shows the number of files. It can be seen that for the CNR level of -5(dB) there were „1000" files in the data directory.

It can also be seen that for levels -5, -3 and -4 ever file was decoded OK, and only 834 „GOOD DECODES" were possible at CNR level -12.

„NO DECODES" counts files that could not be decoded at all, „BAD DECODES" count files that could be decoded, but the results were different from the originally encoded data. „GOOD SYNCS" show the number of files where the synchronization signal could be recovered with utmost precision. „BAD SYNCS" counts ill-identified sync vectors. „BAD SYNC SET" shows all the offsets where a sync vector was found, except the correct 2050 offset.

3.3.8 Generating synchronization vectors

The synchronization vector has to have a special property: it must have low aperiodic autocorrelation. Such vectors are rare, and long vectors are hard to find. The script `syncvec-search.py` tries to find such vectors using gradient descent with random restarts. This script was used design/development time, and can be used again if (when) changing the modulation becomes necessary.

3.3.9 Observing saved audio waveforms and spectra

A script named `scope.py` is provided for graphically displaying waveforms stored in s16 files. The script accepts a file name as a single mandatory positional parameter, and various flags that control what should be displayed. By default a simple waveform is displayed.

The following flags are available:

-h Show help and exit.

-s Show spectrum.

-c Show correlation vector.

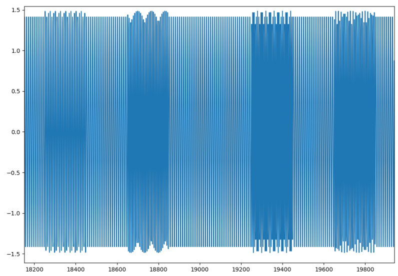
-w Show waterfall.

-o Offset: skip this many samples.

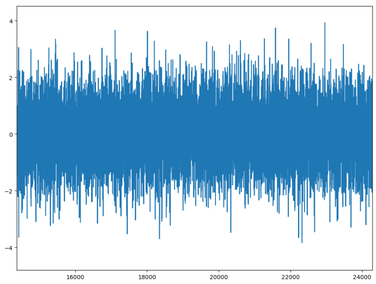
-y Synchronize: try to find a packet, cut it out, and. display only that

Some examples follow showing clean and noisy signals.

```
$ ./scope.py clean.s16
...
$ ./scope.py noisy.s16
...
```

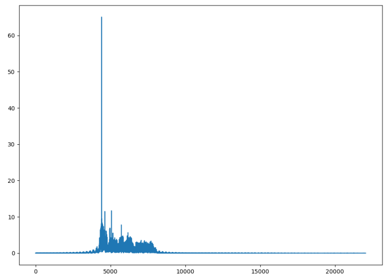


(a) Clean waveform

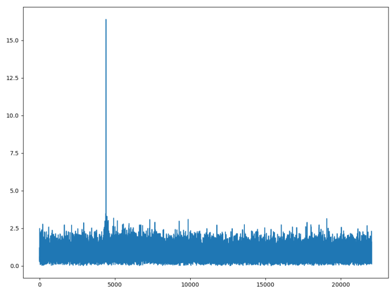


(b) Noisy waveform

```
$ ./scope.py -s clean.s16
...
$ ./scope.py -s noisy.s16
...
```

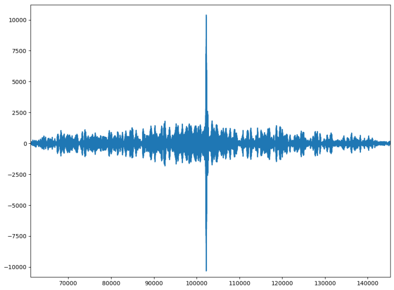


(a) Clean DFT

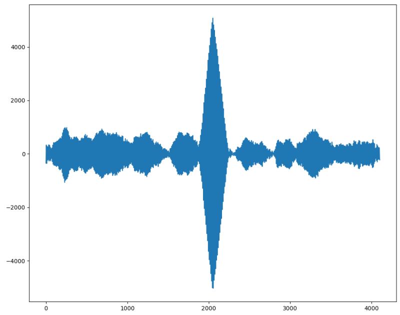


(b) Noisy DFT

```
$ ./scope.py -c clean.s16
...
$ ./scope.py -c noisy.s16
...
```

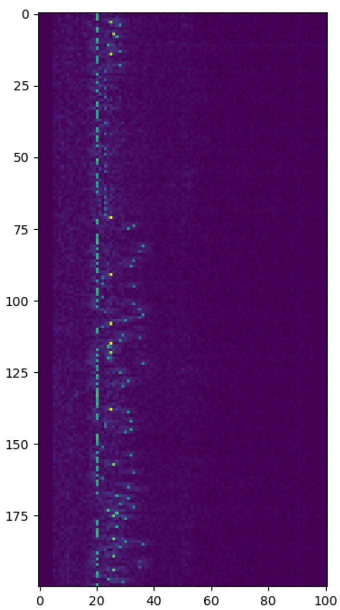


(a) Correlation vector, clean signal

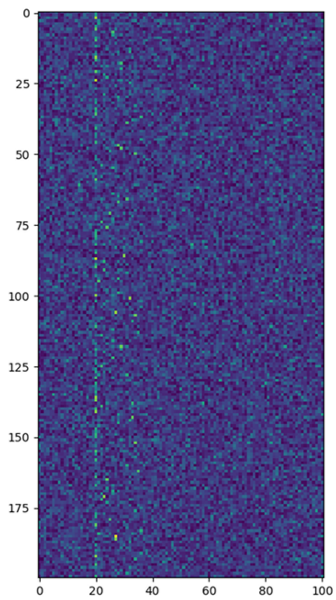


(b) Correlation vector, noisy signal

```
$ ./scope.py -y -w clean.s16
...
$ ./scope.py -y -w noisy.s16
...
```



(a) Clean signal, synchronized



(b) Noisy signal, synchronized

4 Experimental results

A large number of simulations, and some real-world experiments were conducted. These are described below.

4.1 Real-World Experiments

Real-world experiments in the given time frame could only yield a modest number of data points. The following description focuses more on usability and other practical aspects, and does not try to communicate precise figures.

4.1.1 Single PC setup

A single desktop personal computer with a high quality speaker and microphone were used to test performance at the early stages of development.

With this setup I could easily achieve reliable data transfer at distances slightly over 1 meter even when music played, or people were talking if the microphone and speaker were facing each other.

Performance was heavily dependent on the microphone setup and volume setting. Too high volume causes clipping and destroys the signal, to low volume hides the signal in the noise of the recording equipment. If the microphone were facing in a direction other than the speaker, *multipath propagation* was found to interfere with reception.

4.1.2 Single notebook setup

A single laptop computer was used to try and replicate the results. The notebook had a built-in microphone with lots of noise, since it was equipped with forced air cooling, and the fan was on most of the time.

The built-in microphone and the speaker were about 12cm close together. The received signals weren't as high quality as with the PC setup. This was due not to the distance, but to the fact that the speakers and the microphone were deliberately placed and insulated so that the audio feedback is as little as possible.

The microphone in this setup faces away from the speakers, and picks other sounds up easily, including the sound of the speakers bouncing off objects, and arriving from the front. Because of this, the signal gets back to the receiver multiple times with some delay. This can easily confuse the demodulator.

4.1.3 Notebook-PC setup

To rule out any problem with the laptop's microphone and audio circuitry, several experiments were conducted with the PC set being the transmitter, and it being the receiver.

A performance similar to the PC-only setup were measured: signals could easily be picked up from about 1 meter even when the laptop's fan were on, or when some music or noise could be heard in the room. The experiments were repeated with reversed roles, and performance remained similar.

4.1.4 Conclusions

Experiments showed that the software worked beyond expectations, but also revealed performance bottlenecks. The sensitivity to multipath in particular needs to be addressed. Ambient noise will of course interfere with the reception, but the volume of the transmitted signal could always be set high enough to ensure success, and not be particularly annoying (although this is highly subjective, and should carefully be considered while developing the product). Multipath in contrast mostly dependent on the paths, and not the volume, so increasing volume it has little benefit in combating multipath.

4.2 Benchmarks

Using the `test-gen.py` script, 14000 audio files were generated, from and including -14dB CNR to and including -1dB CNR. The files contained randomly generated binary data. All the files were exactly 1 second long, and every file had the encoded data right in the middle, at offset 2050. The decoding benchmark does not „know" about this convention, and has to guess the position correctly in every file.

It was found the decoding is possible 99.9% of the time at -11dB CNR under 22.05 KHz band-limited white noise. Performance were quickly falling under that: about 83.4% good decodes were logged at -12dB, and none at -13db.

It is also worth mentioning that no false decodes were ever observed. The messages either decoded correctly, or didn't at all.

The synchronization method was found out to have been a very good choice, might even have been an overkill. **In the vast majority of the cases, the positions of the data frames were precisely identified, even in cases where decoding was impossible.** Even when the algorithm made errors finding frames, it only missed 10 samples in either direction, so instead of one, only 3 positions were ever identified. This is a useful property, and worth noting it in case the synchronization precision would need to be improved for some reason.

5 Implementation Challenges, Possible Improvements

The work I've done on this project so far is rather experimental. The finished product has to work in different environments, with various noise sources, and a large variety of devices with different properties.

I hereby attempt to address a few problems that might come up during the deployment of this – or similar – solution.

5.1 Parameters, Tuning

The performance of the components of the system is not well balanced. The synchronization works very well, but also takes about the half of the bandwidth away. It's possible to make a trade-off between the effectiveness of the synchronization algorithm and the bandwidth it uses. Bandwidth then can be used to increase symbol time and combat noise and multipath, and/or to decrease code rate, and have more immunity to momentarily signal dropout and similar burst errors.

There is a trade-off between the code rate and the symbol time even when we consider the sync vector to be constant. A longer symbol time and higher code rate might provide better white noise and multipath immunity. The possible set of parameters can be explored automatically with machine learning algorithms, similarly to how good synchronization vectors were found in the first place.

5.2 Processing power

In the current form, the software consumes quite a lot of CPU. The implementation of the decoding algorithm must be chosen carefully, and well-optimized native code will probably be needed for acceptable performance.

5.3 Channel model

The simulations use a simple Additive Gaussian White Noise channel. Real-world noise will very likely to be different. Any further development would benefit from a noise model that is more faithful to the environments in which the solution will be deployed.

5.4 Choice of FEC, Implementation, Performance

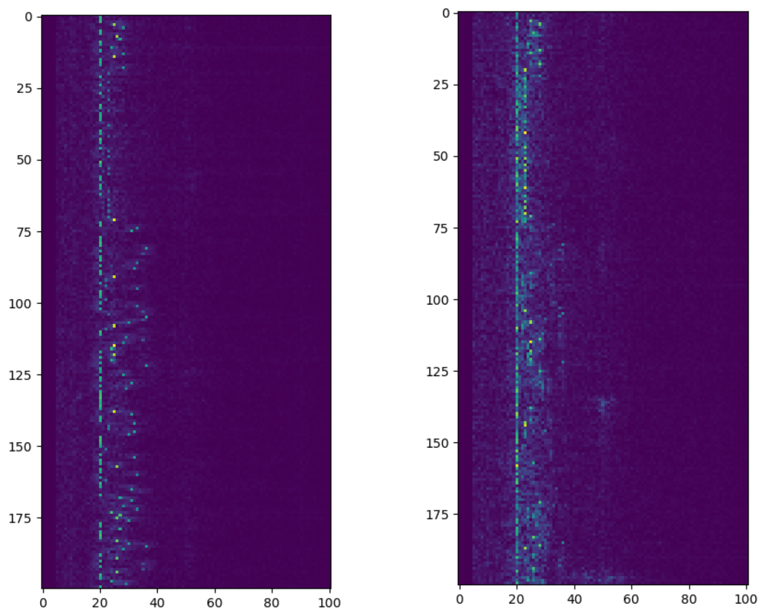
The Franke-Taylor algorithm[3] can be adapted for RS(54,20). This could provide as much as 2dB improvement over the hard-decision algebraic decoding that were used in the software.

Other modern codes, like LDPC codes could be used instead of Reed-Solomon codes. Because the lack of ready-to-use implementations, using such codes was not attempted.

5.5 Multipath

Currently this is one of the major factors that limit the performance of this solution. It can confuse the receiver even when noise and distance are not problems. It is also not obvious what is happening, and multipath can cause seemingly random, frustrating packet loss.

On the following waterfall charts a clean and a multipath-affected signal can be seen:

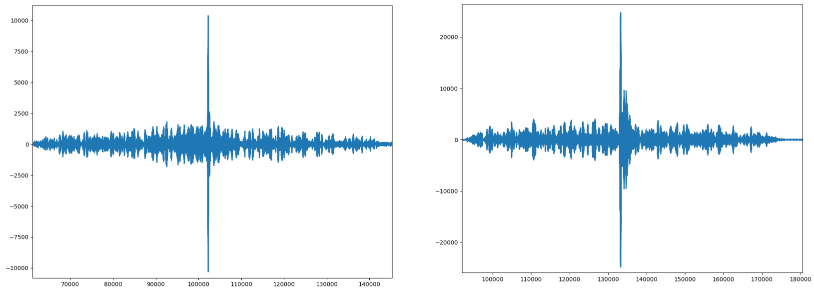


(a) Clean signal

(b) Multipath-affected signal

Notice how the power creeps from one symbol to the next few one, and blur the image, making correct demodulation impossible. This signal cannot be decoded, despite having no significant noise in it.

The next images show correlation vectors of the same signals. The correlation vector contains information about the position of the received frame. The clean signal has a sharp, single spike with small side lobes, whereas the multipath-affected signal has multiple large bumps near the main spike that also looks flattened. This is a sign of multiple synchronization vectors in the same audio. During the experiment only one transmitter was used, so these must be different copies of the very same signal bouncing around the room, and getting back into the microphone.



(a) Correlation vector of a reasonably clean signal (b) Correlation vector of a signal affected by multipath

This problem were almost only encountered when the transmitting speaker and the receiving microphone weren't facing each other. Microphones and speakers tend to be directional *by design*. If they are facing each other, they amplify the signal, and signals coming from other directions are attenuated. If they are facing away each other, stray signals are amplified, and the original one gets attenuated.

This phenomenon can be mitigated by increasing the symbol time, and even by introducing guard intervals. This decreases symbol rate, and makes synchronization harder. Using frequency hopping spread spectrum (FHSS) is an other effective way to combat this issue.

To tackle this problem efficiently, more information on the transmitting equipment and the working environments is needed, because the carrier frequency, bandwidth, noise, and multipath models need to be gotten right for the design of the system to be optimal.

References

[1] SoX, `http://sox.sourceforge.net/`

[2] Anaconda, `https://www.anaconda.com/download`

[3] Steven J. Franke, K9AN and Joseph H. Taylor, K1JT *Open Source Soft-Decision Decoder for the JT65 (63,12) Reed-Solomon Code*, QEX, 2016