TECHNISCHE
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# ANSIAN - ANDROID SIGNAL ANALYZER

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Secure Mobile Networking Lab
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SEEMOO
SECURE MOBILE NETWORKING

AnSiAn - Android Signal Analyzer
SEEMOO Secure Networking Lab

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# ABSTRACT

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# ACRONYMS

| | |
|---|---|
| ADC | Analog-to-Digital Converter |
| AGC | Automatic Gain Control |
| AM | Amplitude Modulation |
| AnSiAn | Android Signal Analyzer |
| BPSK | Binary Phase Shift Keying |
| FFT | Fast Fourier Transform |
| FM | Frequency Modulation |
| GC | Garbage Collector |
| GUI | Graphical User Interface |
| IF | Intermediate Frequency |
| MVC | Model–View–Controller |
| PI | Program Identifier |
| PLL | Phase Locked Loop |
| PSK31 | Phase Shift Keying, 31 Baud |
| RDS | Radio Data System |
| RF | Radio Frequency |
| SDR | Software-Defined Radio |
| SSB | Single Side Band |
| USB | Upper Side Band |

# 1 INTRODUCTION

Android Signal Analyzer (AnSiAn) is an Android application developed by the Secure Mobile Network Lab (SEEMOO) at Technische Universität Darmstadt. It allows for common Software-Defined Radio (SDR) platforms, such as the HackRF and the RTL-SDR, to be used with Android devices. This enables a user to inconspicuously sniff and analyze wireless signals on the go.

The project is based on the open-source app RFAnalyzer by Dennis Mantz [2], which features visual browsing through the frequency domain and demodulation of e.g. Amplitude Modulation (AM) and Frequency Modulation (FM) signals. It has been further developed by Markus Grau and Steffen Kreis into what is now AnSiAn in 2015.

To date, AnSiAn adds to following features on top of RFAnalyzer:

- Waveform graph of received signalsu
- Morse Demodulation and Decoding
- RF Scanner that allows for visually scanning a broad spectrum for signals
- Restructured Graphical User Interface (GUI) with better usability
- Restructured codebase that follows the Model–View–Controller (MVC) pattern

While this makes AnSiAn a powerful tool for mobile signal analysis and processing, further features such as support for additional modulation techniques are desirable. This lab aims to further develop AnSiAn in order to extend its feature set, improve app stability and refine existing features.

A detailed descrpition and explanation of the project goals and their planned schedule is given in Chapter 2. Chapter 3 covers the actual project progress over time as well as encountered problems, delays and unplanned features in each phase of the project. Details on the design and implementation of features and bugfixes are given in Chapter 4. Chapter 5 concludes this documentation.

# 2 PROJECT DEFINITION

This chapter defines the mandatory and optional features that were scheduled for implementation throughout the project and divides them into three sprints.

## 2.1 FEATURES

The new features, that were to be implemented, can be divided into two groups: mandatory features of high priority within this project and optional features, that were to be implemented if time permitted. As can be seen in Section 2.2, the third sprint was reserved either for the implementation of optional features or for completing mandatory features and this documentation.

### 2.1.1 *Mandatory Features*

The following core features were scheduled for implementation during the first and second sprint:

- Radio Data System (RDS) demodulation
  If the user selects the existing wide-band FM demodulation option, the app shall try to detect and demodulate any existing RDS signal along with the FM audio demodulation. The extracted information shall be displayed on the screen.

- Phase Shift Keying, 31 Baud (PSK31) demodulation
  If the user selects Upper Side Band (USB) demodulation mode, he or she shall have the option to enable PSK31 demodulation along with the audio demodulation. The demodulated text string shall appear and scroll through the analyzer window.

- Extraction of demodulated RDS-, Morse- and PSK31-data to logfiles
  If the user selects to demodulate any digital modulation, the demodulated text shall be written to a log file specified by the user.

- Support for the rad1o badge
  The rad1o badge, which is a modified low-cost replica of the HackRF, shall be supported as a signal source by AnSiAn.

- Transmission support for HackRF and rad1o
  If AnSiAn is used with an SDR capable of transmitting signals, it shall offer options to send signals in the following ways:

    - Replay I/O samples from a file
    - Generate and send Morse code from text
    - FM-modulate and send audio from a file

### 2.1.2 *Optional Features*

Optional features were scheduled for the third and last sprint. However, they were only to be implemented if the last sprint was not needed in order to compensate for delays on the mandatory features. The optional features are listed in order of priority:

- Walkie-Talkie Mode
  The user shall have the possibility to put AnSiAn into a Walkie-Talkie mode. In this mode, the application will demodulate an FM channel and the user can quickly switch between demodulation and transmission of audio recorded from the internal microphone.

- Packet Radio demodulation
  A new mode *Packet Radio* shall be added to AnSiAn. Once selected, it shall allow the user to tune to a Packet Radio channel and display information about demodulated packets on the screen. If time permits, it might even be possible to implement a transmission feature for Packet Radio.

## 2.2 TIME SCHEDULE

The project had two developers, Dennis Mantz and Max Engelhardt, working in three sprints. There were three milestones corresponding to the sprints, labeled Alpha, Beta and Final Version. They each add an independent and self-contained set of features to the application:

- Software Design (due 12.05.)
- Sprint 1: Alpha Version (due 09.06.)
  - RDS demodulation
  - PSK31 demodulation
  - Extraction of RDS-, Morse- and PSK31-data to logfiles
- Sprint 2: Beta Version (due 21.07.)
  - Support for the rad1o badge
  - Transmission support for HackRF and rad1o
    - Replay I/O samples from a file
    - Generate and send Morse code from text
    - FM-modulate and send audio from a file

- Sprint 3: Final Version (due 25.08.)
  - Complete leftovers from previous sprints
  - Walkie-Talkie Mode (optional)
  - Packet Radio demodulation (optional)

# 3 PROJECT PROGRESS

## 3.1 SPRINT 1: ALPHA VERSION

## 3.2 SPRINT 2: BETA VERSION

## 3.3 SPRINT 3: FINAL VERSION

# 4 DESIGN AND IMPLEMENTATION

This chapter covers the design and implementation of the new features. It also addresses structural changes, that were necessary for these features to be implemented, as well as optimizations and bugfixes in the original codebase.

## 4.1 MEMORY OPTIMIZATIONS

The original RF Analyzer application's architecture is based on blocking queues that synchronize the various signal processing threads and efficiently manage memory buffers. Unfortunately, this architecture was partly dropped by the developers of AnSiAn when changing to a new architecture based on the EventBus library. As a result, memory allocation management does not work as efficiently with the current version of AnSiAn.

Instead of using cycling buffers for inter-thread-communication, AnSiAn uses EventBus to deliver data. Buffers are always allocated freshly and discarded after use. This results in a high activity of the Garbage Collector (GC) and therefore in a bad overall performance of the app.

Listing 1 shows a logcat output of the app before any optimizations were applied. The GC runs approximately 8 times per second and the slow performance results in stuttering audio demodulation on older hardware.

Listing 1: Logcat output before memory optimizations

```
05-12 17:55:04.060 D/dalvikvm: GC_FOR_ALLOC freed 4347K, 14% free
     54695K/62984K, paused 28ms, total 28ms
05-12 17:55:04.180 D/dalvikvm: GC_FOR_ALLOC freed 4321K, 14% free
     54737K/62984K, paused 26ms, total 26ms
05-12 17:55:04.300 D/dalvikvm: GC_FOR_ALLOC freed 4507K, 14% free
     54705K/62984K, paused 32ms, total 32ms
05-12 17:55:04.420 D/dalvikvm: GC_FOR_ALLOC freed 4454K, 14% free
     54759K/62984K, paused 30ms, total 30ms
```

In order to fix this performance issue, the architecture is reverted to using blocking queues and cycling buffers in places where large memory buffers are passed between threads. EventBus is still used for delivering information which is not tied to large buffers. A schema of the new architecture is depicted in Figure 1.

In this architecture, the buffers cycle between the threads. The reusage of buffers helps to reduce the memory allocation and garbage collection overhead to a minimum. Listing 2 shows the logcat output

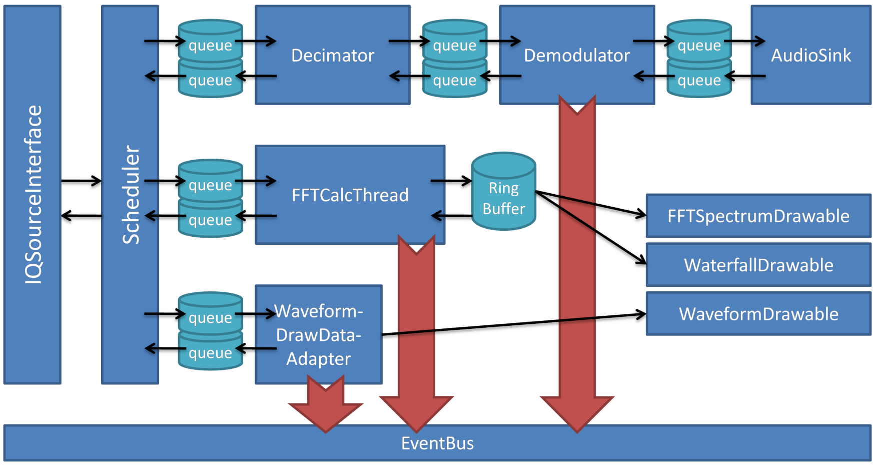


Figure 1: Signal processing architecture with blocking queues

after the architecture changes have been applied. The GC only needs to run every 10 to 20 seconds.

Listing 2: Logcat output after memory optimizations

```
05-12 17:27:29.230 D/dalvikvm: GC_FOR_ALLOC freed 3233K, 15% free
     19706K/23000K, paused 32ms, total 33ms
05-12 17:27:40.780 D/dalvikvm: GC_FOR_ALLOC freed 3528K, 16% free
     20235K/23824K, paused 30ms, total 31ms
05-12 17:28:00.110 D/dalvikvm: GC_FOR_ALLOC freed 4130K, 18% free
     20338K/24528K, paused 36ms, total 37ms
05-12 17:28:24.520 D/dalvikvm: GC_FOR_ALLOC freed 4263K, 18% free
     20341K/24664K, paused 49ms, total 49ms
```

Another significant performance improvement can be gained by synchronising the calculation of the Fast Fourier Transform (FFT) to the refresh cycle of the application GUI. This simple change reduces the number of time the FFT algorithm has to be executed and releases processing resources for other calculations (e.g. in the demodulation path).

Unfortunately, it turned out that the Morse demodulation algorithm of the previous version of AnSiAn was working in the frequency domain and relied on continuous FFT samples without gaps. As result this performance improvement comes at the cost of reimplementing the Morse demodulation algorithm in order to work on time domain samples instead.

## 4.2 DEMODULATORS

This section describes the design and implementation process of the developed demodulators. Section 4.2.1 covers where and how the new RDS and PSK31 demodulators were integrated into the existing architecture and which changes were necessary. Due to performance issues with the old frequency-domain-based Morse demodulator, it had to be rewritten. This process is described in . The implementation of the new RDS and PSK31 demodulators is coverd in Sections 4.2.3 and 4.2.4 respectively.

### 4.2.1 *Design and Structural Changes*

The existing architecture of AnSiAn features individual threads for scheduling, downsampling, demodulation and audio output. The `Demodulator` thread demodulates quadrature samples by calling the `demodulate()` method on an instance of `Demodulation`. `Demodulation` is an abstract class that is implemented by concrete demodulation methods such as `AM`, `FM` and `Morse`.

AnSiAn utilizes the EventBus library in order to pass demodulated Morse text to the GUI. Demodulated audio data is passed to the `AudioSink` thread by enqueuing it into its input queue. This mechanism is explained in more detail in Section 4.1.

In order to extend AnSiAn with demodulation functionality for PSK31 and RDS, the existing architecture needs to be extended by two new subclasses. The extended architecture is depicted in Figure 2 and explained in the following.

Two new classes `PSK31` and `RDS` are integrated into the existing demodulation classes for USB and FM. PSK31 demodulation works on the envelope of the received USB signal. RDS transmits metadata for FM radio channels, which is why the RDS demodulation needs to work on an intermediate result of the wide-band FM demodulation class. Through a new option in the app settings it will be possible to enable or disable the demodulation of the embedded digital modes when the corresponding analog mode is selected.

Like the existing architecture, the new architecture will use the EventBus library to pass the demodulated text to the GUI. The existing View `MorseReceiveView` is refactored into a universal `DemodulationInfoView` that displays the text output of any selected demodulator. Demodulators pass `DemodInfoEvents` via the EventBus to the `DemodulationInfoView`, which contain demodulated text and further information (e.g. baud rates or raw dits and dahs) and are displayed in separate lines.

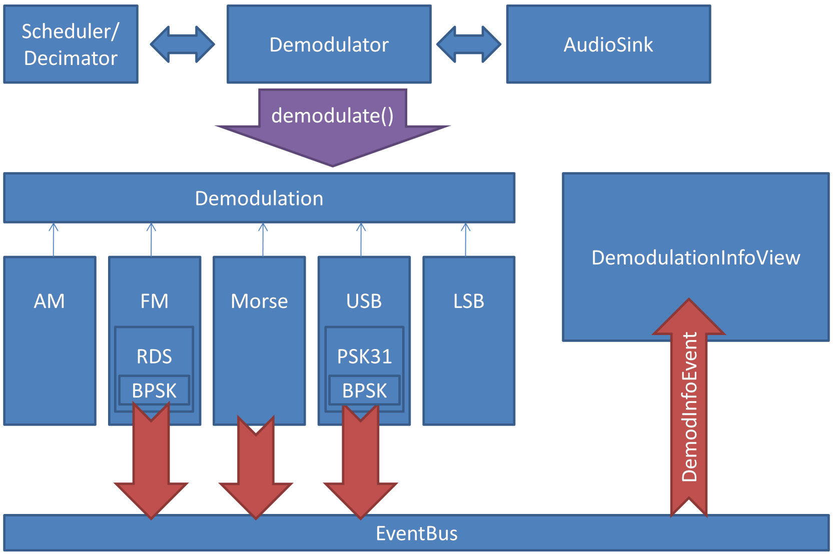


Figure 2: Architecture of the extended demodulation logic and communication with the GUI

### 4.2.2 *Re-Implementation of Morse Demodulator*

As explained in Section 4.1, the original version of AnSiAn calculated FFTs excessively often and thus offered bad performance. AnSiAn's original Morse demodulator, which operated in the frequency domain, relied on frequent FFT calculations and did not work after the optimizatons described in Section 4.1. In order to still demodulate Morse with the new efficient architecture, the Morse demodulator needed to be re-written to work in the time domain rather than the computationally expensive frequency domain.

#### 4.2.2.1 *Requirements*

Aside from demodulating and decoding Morse transmissions, the old demodulator offered support for the following features, which the new demodulator should implement as well.

- "Manual", "semi-automatic" or "automatic" detection of dit duration and threshold
- Optional automatic re-initialization of detected dit duration and threshold in case of high error rate
- Optional AM audio demodulation along with Morse demodulation

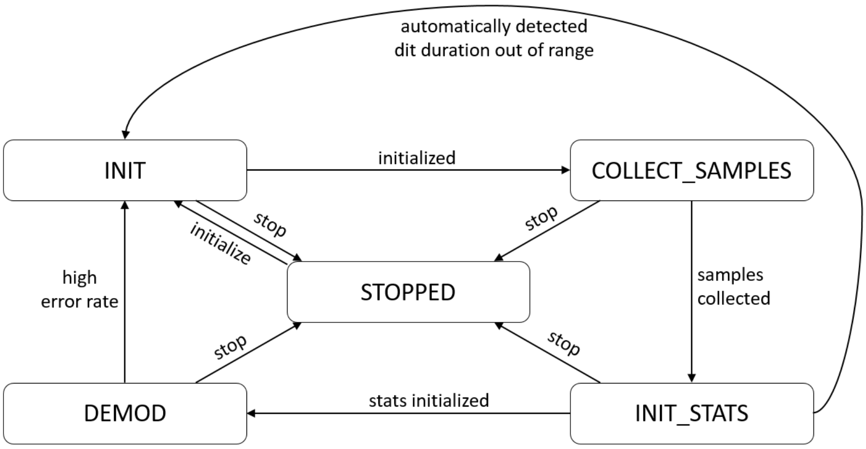


Figure 3: States and state transitions of the re-written Morse demodulator

In automatic mode, the demodulor automatically estimates the high-/low threshold and the duration of a dit based on a set of samples, that needs to be collected before the actual demodulation can start. Manual mode requires the user to explicitly specify the dit duration and uses the current squelch as threshold. In semi-automatic mode, the user specifies the dit duration while the threshold is determined automatically.

With the new demodulator getting its samples from the `Decimator` rather than the `FFTCalcThread`, the practice of using the squelch as threshold is no longer applicable, as samples below the squelch level do not get passed to the `Demodulator`. The new implementation of the Morse demodulator thus only supports "automatic" and "manual" mode with respect to dit duration. The threshold is always determined automatically.

#### 4.2.2.2 *Implementation*

The re-written Morse demodulator has five states that are displayed in Figure 3 and explained in the following.

INIT In this state, the demodulator initializes its internal state. Samples received while the demodulator is in this state, are discarded and not demodulated (in case AM demodulation is activated, audio still gets demodulated).

COLLECT_SAMPLES The demodulator calculates and stores the envelope of all received samples in a buffer. In case AM demodulation is activated, audio gets demodulated.

INIT_STATS The collected samples from the previous state are used to initialize the threshold and, optionally, the dit duration. Samples received while the demodulator is in this state, are discarded and not demodulated (in case AM demodulation is activated, audio still gets demodulated).

DEMOD Received samples are demodulated on-the-fly. If demodulation or decoding error rates are beyond a threshold, the demodulator re-initializes.

STOPPED The demodulator should not be running; all samples are discarded.

For envelope detection, $\sqrt{I^2 + Q^2}$ is used, with I and Q being the inphase and quadrature components of the sample, respectively. This is computationally more expensive than using the existing AM demodulator to detect the envelope, but produces more exact results and comes without the unwanted automatic gain control of the AM demodulator. For efficient memory management, the envelope of the currently processed sample packet is not always stored in a new array. Instead, a buffer is allocated once and overwritten with the current envelope each time a new sample packet is processed. If AM demodulation is enabled, the detected envelope is copied from the buffer to the output-packet (which is passed to the `AudioSink`).

The threshold, that determines whether a given sample is a "low" or a "high" sample (i.e. if it is part of a dit/dah or a pause), is initialized based on the initialization data collected during the COLLECT_SAMPLES step and continuously updated for every processed sample. It is calculated as $bottom + \frac{peak - bottom}{2}$, with $bottom$ being the lowest envelope and $peak$ being the highest envelope seen since the demodulator's initialization.

If automatic mode is enabled, the dit duration is determined by counting streaks of consecutive "high" (i.e. bigger than the threshold) samples in the initialization data. The observed streak lengths are then adjusted by adding the number of observed streaks with one sample more or less (i.e. $adjustedo_{new}(x) = o(x-1) + o(x) + o(x+1)$ with $o(x)$ being the number of occurences of a continuous high-streak of $x$ samples). This is done to ignore outliers. The most-frequently-observed streak length is assumed to be the number of samples per dit.

In DEMOD state, streaks of continuous high and low values are counted in the currently processed samples packets. The length of a streak and its value (i.e. high or low) are then decoded to find out whether it encodes a dit, dah or a short, medium or long pause. If a medium pause (encodes the end of character symbol) is detected, the previous dits and dahs are passed to the `Decoder` class, which translates them into a character. The `Decoder` class was already used by the old demodulator, originally taken from the open-source project Morsecoder and could be recycled for the new demodulator.

To implement automatic re-initialization, the demodulator uses `ErrorBitSets`, which are essentially ring buffers for boolean values. Two `ErrorBitSets` are used; one for demodulation (i.e. translating streak lengths into dits, dahs and pauses) and one for decoding (i.e. translating dits and dahs into characters). Every entry represents a successful or unsuccessful decoding/demodulation attempt. The size of the `ErrorBitSet` that tracks demodulation success is 100, the size for the decoding success `ErrorBitSet` is 30. If the success rate drops below 50%, the demodulator re-initializes itselt and re-determines the threshold and, if automatic mode is enabled, dit duration.

Listing 3 shows an excerpt of the demodulator's core method `demodulate(SamplePacket input, SamplePacket output)`. The operation of the demodulator, which was described in this section, can be recognized in it.

Listing 3: The demodulate method of the new Morse demodulator

```
switch (this.state) {
    case INIT:
        if (amDemod) {
            // do nothing except AM Demodulation;
                Demodulator might be in inconsistent
                state
            envelopeToBuffer(input);
            amDemodFromBuffer(output);
        }
        break;
    case COLLECT_SAMPLES:
        // collect samples and write them to initSamples
            until we have enough
        envelopeToBuffer(input);
        if (amDemod)
            amDemodFromBuffer(output);
        collectSamplesFromBuffer();
        if (!(initSamplesCollected <
            initSamplesRequiredForInit)) {
            initializeStats();
        }
        break;
    case INIT_STATS:
        if (amDemod) {
            // do nothing except AM Demodulation;
                Demodulator might be in inconsistent
                state
            envelopeToBuffer(input);
            amDemodFromBuffer(output);
        }
        break;
    case DEMOD:
        // demodulate samples
        envelopeToBuffer(input);
        if (amDemod)
```

```
            amDemodFromBuffer(output);
        updateThresholdFromBuffer();
        binarizeBuffer();
        demodulateBuffer();
        if (automaticReinit && needsReinit()) {
            MyToast.makeText("High decoding error rate;
                reinitializing...", Toast.LENGTH_LONG);
            init();
        }
        break;
    case STOPPED:
        // discard samples; Demodulator should not be
            running
        break;
}
```

### 4.2.3 *Radio Data System*

The RDS signal is transmitted along with wide band FM radio signals to provide additional information about the radio station and program.

Demodulation of the RDS signal is first done in Octave in order to evaluate the demodulation algorithm. The octave implementation also helps by providing reference data of the different stages of demodulation.

#### 4.2.3.1 *RDS modulation scheme*

RDS uses Binary Phase Shift Keying (BPSK) with Manchester encoding. The signal is transmitted with an offset of 57 kHz relative to the center frequency of the mono audio signal (baseband). The 19 kHz pilot tone of wideband FM can therefore be used to retrieve the RDS carrier by multiplying it with itself 3 times. The complete FM spectrum can be seen in Figure 7a.

After the RDS baseband signal has been retrieved from the FM signal there are multiple ways of demodulating the BPSK modulation. A sophisticated approach tries to recover the phase synchronised RDS carrier from the signal by using e.g. a form of Phase Locked Loop (PLL) or Costas Loop. The symbols can then be extracted by multiplying the carrier with the modulated signal and apply a threshold operation to get bits.

A much simpler approach is to analyze the envelope of the signal and dectect bits based on known shapes of ones and zeros in the waveform. Figure 4 shows the envelope and the pattern of symbols which can be detected.

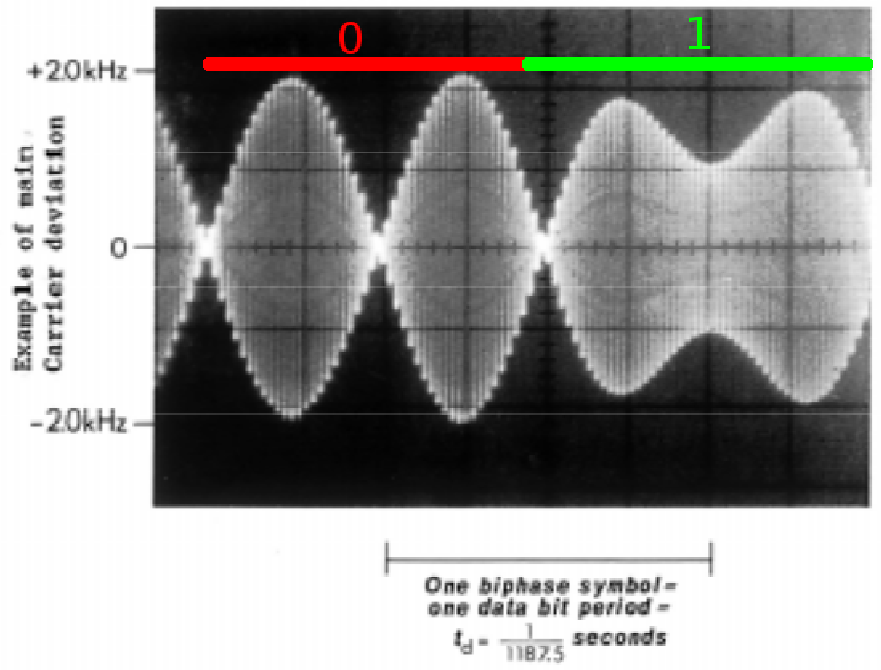


Figure 4: RDS envelope waveform after Frequency demodulation [1]

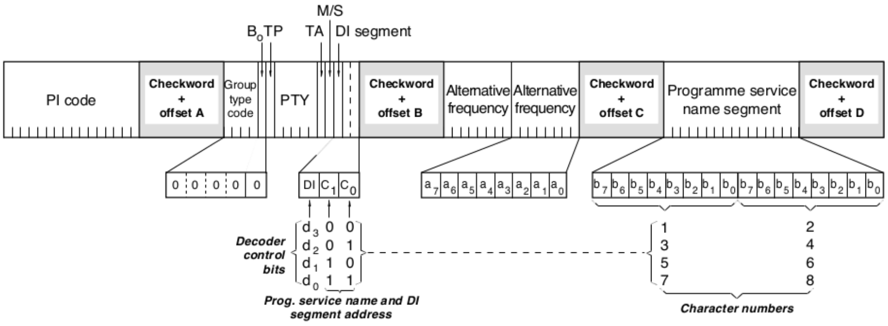


Figure 5: Coding scheme of RDS: group 0A [1]

#### 4.2.3.2 *RDS coding scheme*

RDS frames are called groups and each group consists of 4 blocks called A, B, C and D. One block has a length of 16 bit plus a 10 bit checkword. Block A always contains the Program Identifier (PI) which identifies the radio station. The content of the other blocks depends on the group type which is located in block B (see Figure 5).

Table 4 lists all group types and their descriptions. The RDS demodulation in AnSiAn only decodes types 0 and 2 because they contain the basic information which is also often displayed on the radio receiver.

#### 4.2.3.3 *Evaluation in Octave*

Developing a signal processing application on Android has many drawbacks. One issue is that it is very hard to debug the actual signal processing components because of the lack of proper tools to visualize and analyze the data that is being processed. It is also not possible to do rapid prototyping without sufficient signal processing libraries available. Therefore the RDS demodulator was first developed in Octave and afterwards ported to Android.

For development and testing it is better to work on recorded samples instead of live captures. This makes tests reproducible and simplify the development environment. The file was recorded using the record feature of RF Analyzer. It can be imported to Octave by using the `read_cuchar_binary()` script provided by the GNU Radio project. After each step the produced output data can be written back to an *IQ* file in order to use it in the Android application. This way it is possible to develop each component of the demodulation process separately and the output can be visualized on the developing machine.

The demodulation is done in the following steps:

1. Downmixing the radio signal to baseband and filter it (see Figure 6).

2. FM demodulation (see Figure 7a).

3. Downmixing the RDS signal to baseband and filter it (see Figure 7 b and c).

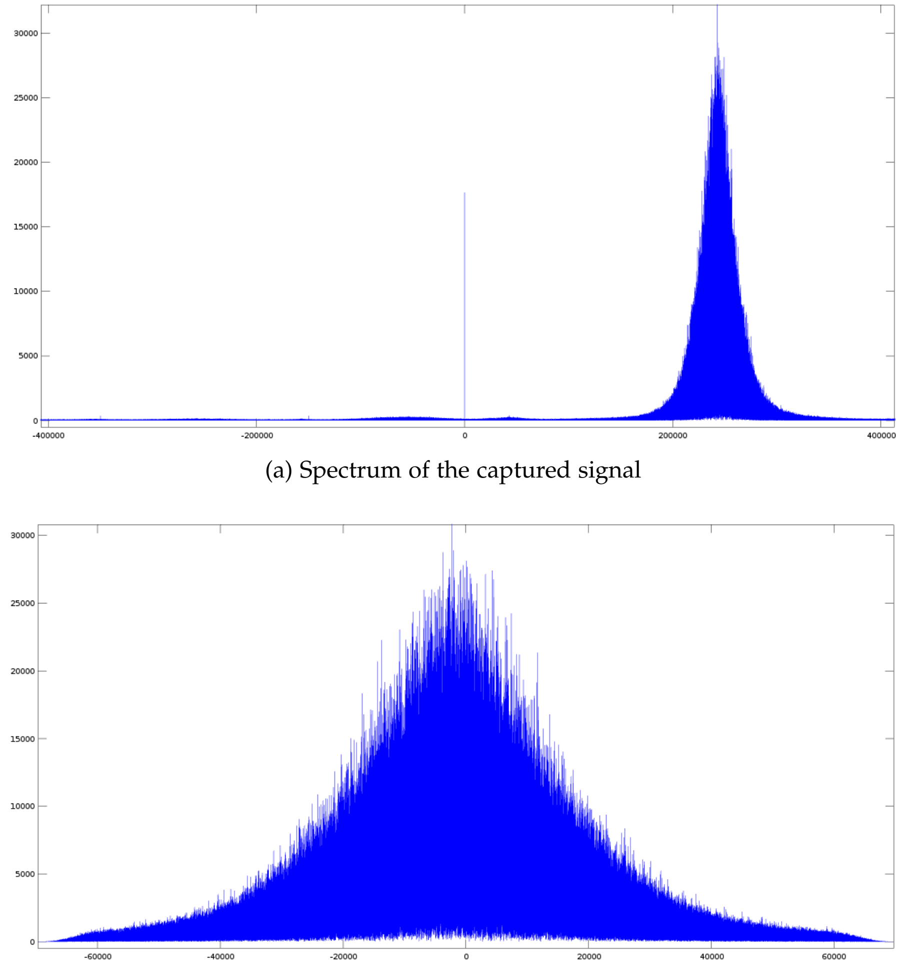
4. Take the absolute value of the signal to get the envelope that was shown above (see Figure 8).

5. Find the beginning of a symbol by searching for a minimum in the waveform. From there find the end of the symbol with the same strategy. Now determine whether the symbol is a one or a zero according to the value of the minimum found in the middle of the sample compared to its peaks.

The octave code used to execute the steps mentioned above is shown in the listing below:

Listing 4: Octave implementation of the RDS demodulator

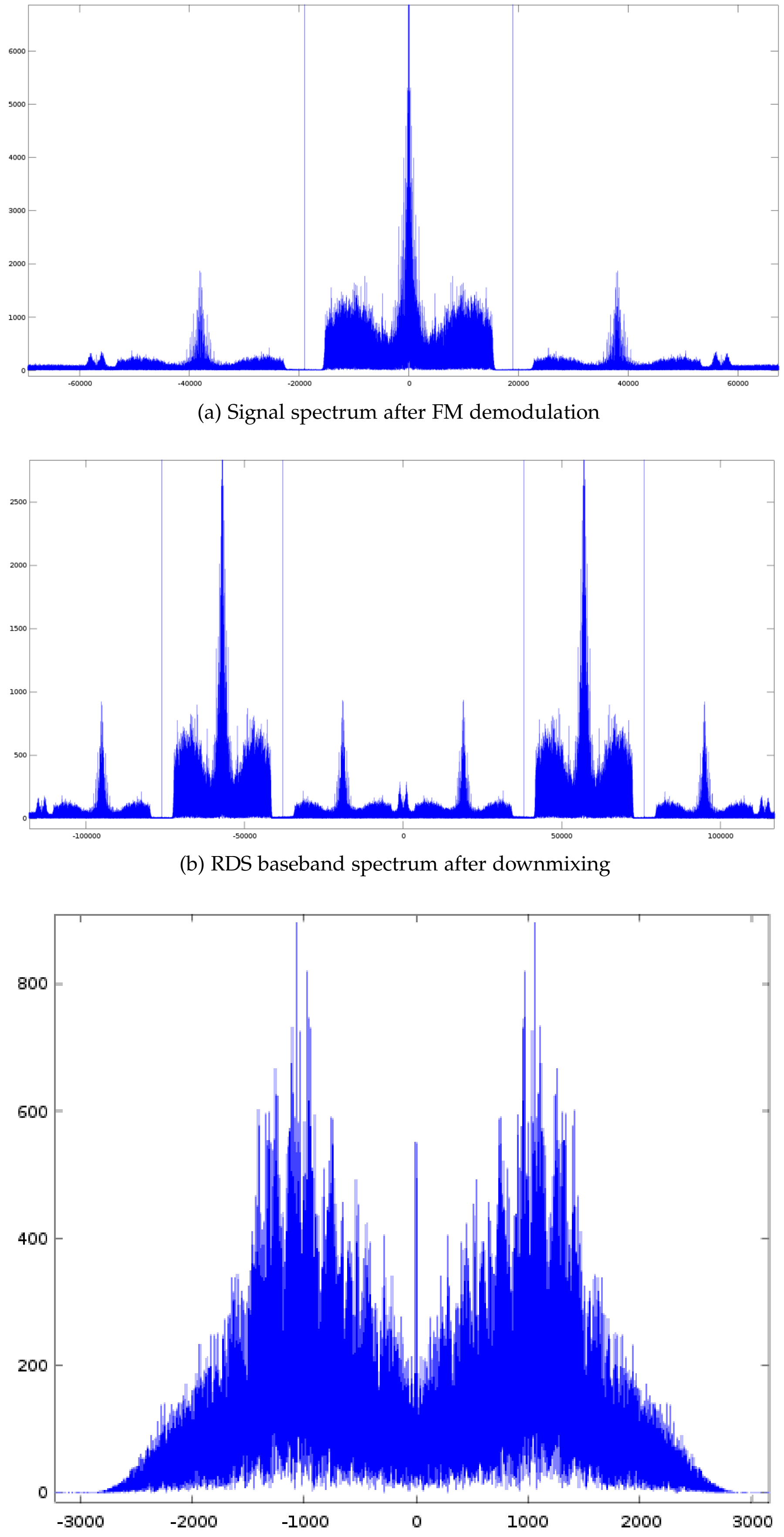
```
signal = read_cuchar_binary ("rtlsdr_100550000Hz_1000000Sps.iq");
t = linspace(0, length(signal)/1000000, length(signal))';
carrier = e.^(2*pi*-245000*t*i);
down = carrier .* signal;
fl = fir1(300, 100000/1000000*2);
filtered = filter(fl, 1, down);
demod = quad_demod(filtered, 1);
t2 = linspace(0, length(demod)/1000000, length(demod));
```



(a) Spectrum of the captured signal

(b) Spectrum after downmixing and filtering

Figure 6: FM Modulated Signal



(a) Signal spectrum after FM demodulation

(b) RDS baseband spectrum after downmixing

(c) RDS baseband spectrum after filtering

Figure 7: Extracting the RDS signal from the FM signal

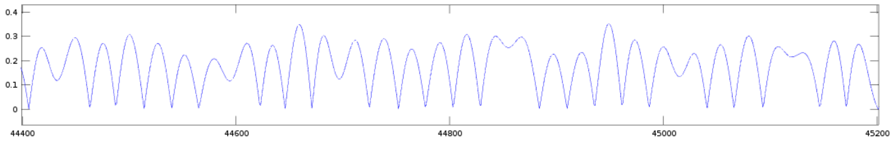


Figure 8: RDS waveform after take the absolute values

```
rdscarrier = cos(2*pi*-57000*t2)';
rdsbase = demod(1:length(rdscarrier)) .* rdscarrier ;
frds = fir1(300, 2400/1000000*2);
rdsbase_filtered = filter(frds,1,rdsbase);
downsampled = decimate(rdsbase_filtered, 16).*80;
write_cuchar_binary (downsampled, "rds_baseband_62500sps.iq");
bits = rds_bpsk_demodulate(downsampled, 62500);
rds_decode(bits)
```

The `quad_demod()` function does the quadrature demodulation (FM demodulation). The `rds_bpsk_demodulate()` function is shown in the following listing:

Listing 5: Octave implementation of the BPSK demodulation

```
function demod = rds_bpsk_demodulate(signal, fs)
  samples_per_symbol = fs/1187.5
  samples_per_symbol = ceil(samples_per_symbol)
  envelope = abs(signal);

  % Find the first minimum
  [minimum, idx1] = min(envelope(1:samples_per_symbol))

  bits = [];
  while (idx1 + samples_per_symbol*2 < length(envelope))
    % find end of symbol idx2 (minimum near idx1 +
        samples_per_symbol)
    from = round(idx1+samples_per_symbol*0.75);
    to   = round(idx1+samples_per_symbol*1.25);
    [minimum, idx2] = min(envelope(from:to));
    idx2 = idx2 + from;

    % calc mean of all samples between idx1 and idx2 and calc
        threshold = mean/2
    m = mean(envelope(idx1:idx2));
    threshold = m/2;

    % get minimum sample in the middle between idx1 and idx2 ...
    span = idx2 - idx1;
    from = round((idx1+idx2)*0.5 - 0.25*span);
    to   = round((idx1+idx2)*0.5 + 0.25*span);
    [minimum, idxmiddle] = min(envelope(from:to));
    idxmiddle = idxmiddle + from;

    % Check whether we have the correct timing. It might be, that
         idx2 is
```

```
   % actually in the middle of a symbol than at its end.
   if (envelope(idx2) > threshold)
    % In this case we find the minimum between idx1 and idx2
      and set it
    % as idx1 for the next round:
    %printf("WARNING: Wrong timing. thres=%f < envelope(idx2=%d
      )=%f\n",threshold,idx2,envelope(idx2));
    idx1 = idxmiddle;
    continue;
   endif

   % ... and check it against the threshold
   s = envelope(idxmiddle);
   if (s > threshold)
    bits = [bits 1];
   else
    bits = [bits 0];
   endif

   % idx1 = idx2 and continue with the next symbol..
   idx1 = idx2;

  endwhile
  demod = bits;
```

#### 4.2.3.4 *Android Implementation*

For the Android implementation two classes are added to the AnSiAn codebase:

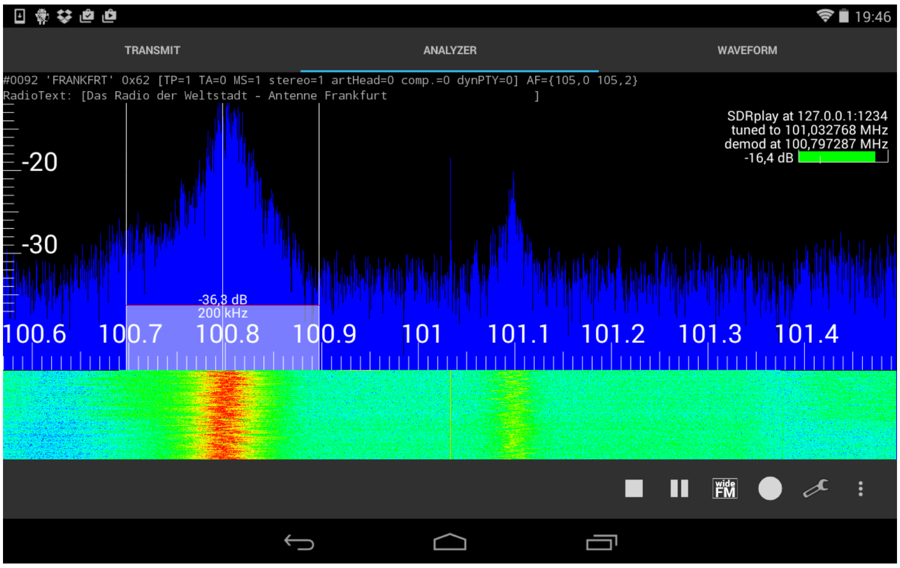
- BPSK: This class handles the BPSK demodulation and can be reused by other demodulators using the BPSK modulation scheme (e.g. PSK31).
- RDS: This class integrates in the existing FM class for frequency demodulation. It handles the decoding and processing of RDS groups.

A screenshot of the application demodulating the RDS signal of the *Antenne Frankfurt* station is shown in Figure 9.

### 4.2.4 *PSK31*

PSK31 refers to phase shift keying modulation using a baud rate of 31.25 Hz. It was developed by Peter Martinez in 1998 to introduce a narrow bandwidth digital mode for live chatting. Because of the simple and efficient modulation and coding scheme (e.g. it does not have an error correction mechanism) it is very widespread amongst amateur radio operators.

A typical setup for PSK31 operation is a Single Side Band (SSB) transceiver connected to the sound card of a computer. The audio



```
#0092 'FRANKFRT' 0x62 [TP=1 TA=0 MS=1 stereo=1 artHead=0 comp.=0 dynPTY=0] AF={105,0 105,2}
RadioText: [Das Radio der Weltstadt - Antenne Frankfurt                    ]
```

```
SDRplay at 127.0.0.1:1234
tuned to 101,032768 MHz
demod at 100,797287 MHz
-16,4 dB
```

Figure 9: Screenshot of the RDS demodulator on a Nexus 7

channel that is fed into the computer can contain multiple parallel PSK31 transmissions, which are demodulated using a special software.

PSK31 is usually operated in USB mode and shall therefore be available in AnSiAn when the USB demodulator is active. As with the RDS demodulator the algorithm to demodulate and decode PSK31 is first evaluated in Octave and later ported to the AnSiAn application.

#### 4.2.4.1 *PSK31 modulation scheme*

PSK31 in its basic form uses BPSK to transmit binary information by sending a single side band signal with either a 180 degree phase shift (digital '1') or without a phase shift (digital '0'). Additionally a root raised cosine filter is used in order to smooth the phase shift and therefore keep the bandwidth narrow. Each symbol contains information about one bit and is always 32 ms long.

Figure 10 shows the modulated PSK31 signal. As already explained in Section 4.2.3.1, the signal may be demodulated by using a second order loop (e.g. a costas loop) in order to recover the phase information and correct for small frequency variations. However the simple approach using envelope detection is also possible. The BPSK code used for RDS demodulation can be partly reused and only needs some modifications as PSK31 does not use Manchester Encoding.

#### 4.2.4.2 *PSK31 coding scheme*

PSK31 uses a variable length encoding called *Varicode* which assigns frequently used characters a shorter code similar to morse. Characters are separated by two consecutive zeros. Table 5 in the appendix lists all characters and their varicode encodings.

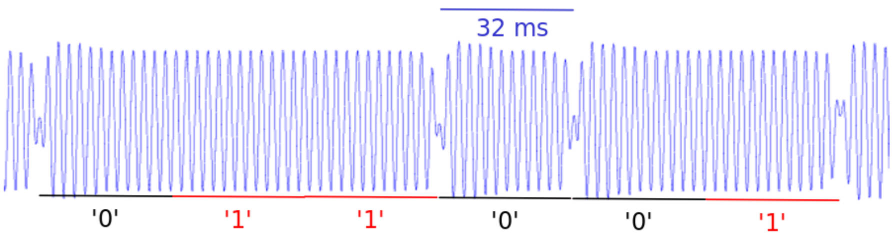


Figure 10: PSK31 modulation scheme: Each symbol is 32ms long. A 180° phase shift indicates a ’0’, no phase shift indicates a ’1’

#### 4.2.4.3 *Evaluation in Octave*

The demodulation algorithm in octave comprises the following steps:

1. Downmixing the USB signal to baseband.
2. USB demodulation (complex bandpass filter to get only the upper side band). See Figure 11 b.
3. Downsampling in order to reduce the workload for the PSK31 demodulation.
4. Envelope detection. See Figure 12.
5. Find the beginning of a ’0’ symbol by searching for a minimum in the envelope of the signal. Determine whether the next symbol is a ’0’ (another minimum is found around 32ms from the beginning of the current symbol) or a ’1’ (no minimum at the beginning of the next symbol). If another ’0’ was detected, the timing can be corrected by centering the minimum at the beginning of the signal and search on from there.

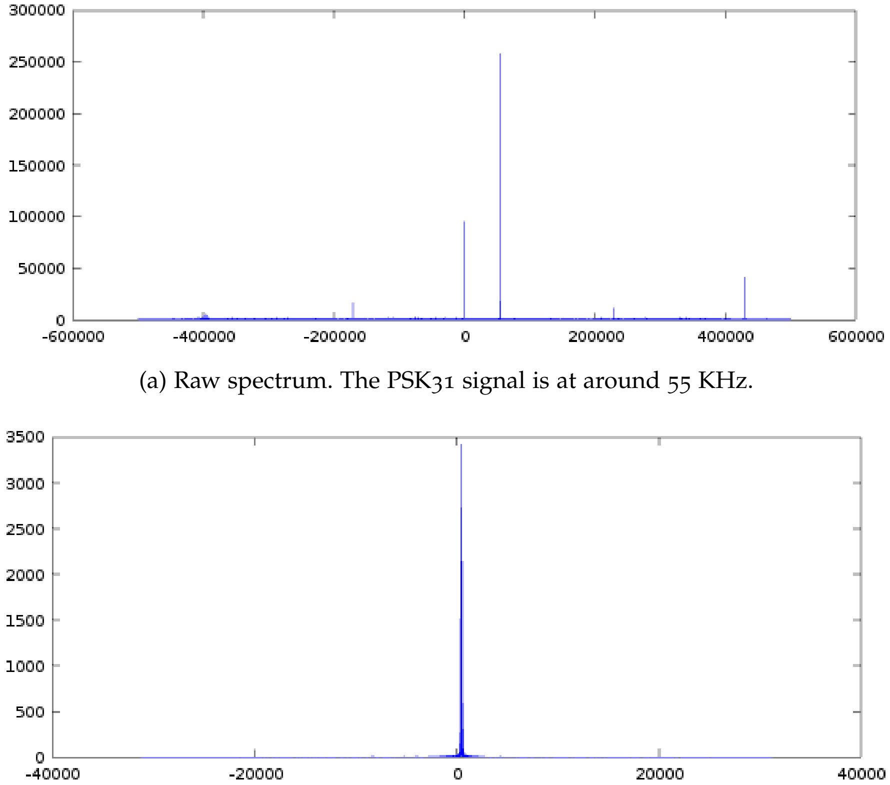
The octave code used to execute the steps mentioned above is shown in the listing below:

Listing 6: Octave implementation of the PSK31 demodulator

```
% load signal
signal = read_cuchar_binary ("rtlsdr_434570917Hz_1000000Sps.iq");
fs = 1000000;

% downmixing (signal is at ~ 55 Khz)
fc = 55000;
t = linspace(0, length(signal)/fs, length(signal))’;
carrier = e.^(2*pi*-fc*t*i);
down = carrier .* signal;

% USB demod (bandpass filter [200Hz-1000Hz])
bpfilter = firls(300, [0 200 200 1000 1000 fs/2]./(fs/2), [0 0 0
    1 0 0]);
filtered = filter(bpfilter, 1, down);
```



(a) Raw spectrum. The PSK31 signal is at around 55 KHz.

(b) Spectrum after downmixing and filtering (USB demodulation)

Figure 11: PSK31 signal in the frequency domain (spectrum)

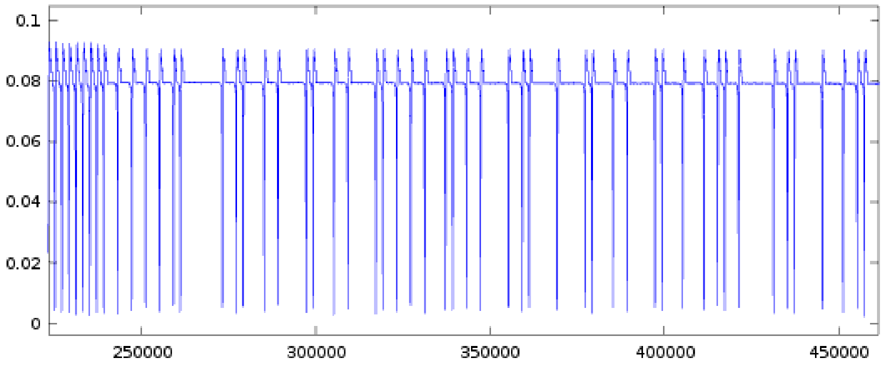


Figure 12: Envelope of the downsampled, USB demodulated signal.

```
% downsampling
downsampled = decimate(filtered, 16);

% BPSK demodulation
bits = psk31_bpsk_demodulate(downsampled, fs/16);

% Varicode decoding
psk31_decode(bits)
```

The `psk31_bpsk_demodulate()` function is shown in the following listing:

Listing 7: Octave implementation of the BPSK demodulation

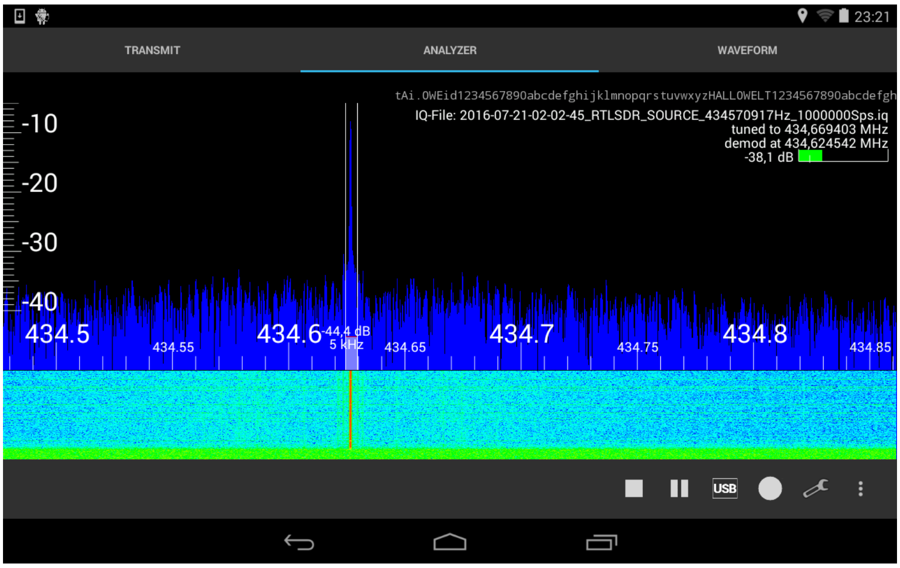
```
function demod = psk31_bpsk_demodulate(signal, fs)
  % PSK31 baud rate is 31.25 Hz
  samples_per_symbol = fs/31.25
  samples_per_symbol = ceil(samples_per_symbol)
  envelope = abs(signal);

  % Find the first minimum
  [minimum, idx1] = min(envelope(1:samples_per_symbol))

  bits = [];
  while (idx1 + samples_per_symbol*2 < length(envelope))
    % search for a minimum at the position of the next sample
    % (minimum near idx1 + samples_per_symbol)
    from = round(idx1+samples_per_symbol*0.5);
    to   = round(idx1+samples_per_symbol*1.5);
    [minimum, idx2] = min(envelope(from:to));
    idx2 = idx2 + from;

    % calc mean of all samples between idx1 and idx2 and calc
        threshold = mean/2
    m = mean(envelope(idx1:idx2));
    threshold = m/2;

    % Check whether we have a minimum (->0) or not (->1).
    if (envelope(idx2) > threshold)
      % In this case we have a bit 1.
      bits = [bits 1];
      idx1 = idx1 + samples_per_symbol;
    else
      % In this case we have a bit 0.
      bits = [bits 0];
      idx1 = idx2;
    endif
  endwhile
  demod = bits;
```



```
tAi.0WEid1234567890abcdefghijklmnopqrstuvwxyzHALLOWELT1234567890abcdefgh
IQ-File: 2016-07-21-02-02-45_RTLSDR_SOURCE_434570917Hz_1000000Sps.iq
tuned to 434,669403 MHz
demod at 434,624542 MHz
-38,1 dB
```

Figure 13: Screenshot of the PSK31 demodulator on a Nexus 7

#### 4.2.4.4 *Android Implementation*

For the Android implementation the `BPSK` class is modified to handle both BPSK variations needed in AnSiAn:

- RDS: BPSK with Manchester Encoding
- PSK31: BPSK without Manchester Encoding

Additionally the class `PSK31` is added which is a subclass of `Demodulation`. It integrates with the existing `USB` class to provide PSK31 demodulation when USB demodulation is active.

A screenshot of the application demodulating a PSK31 signal is shown in Figure 13.

## 4.3 GUI

### 4.3.1 *Reorganization of Preferences*

### 4.3.2 *Transmit Tab*

## 4.4 SUPPORT FOR NEW SDR PLATFORMS

AnSiAn supports different SDR hardware such as:

- RTL-SDR dongle
- HackRF

The architecture of the app allows to easily add support for additional hardware, as long as it provides IQ data samples and an Android driver is available.

### 4.4.1 *rad10*

The *rad10* is a SDR platform with the same design as the *HackRF* that was used as badge for the *Chaos Communication Camp* 2015. Some modifications to the design were made in order to reduce costs and therefore minor differences exist compared to a HackRF (e.g. the option to activate the antenna power is missing for the *rad10*).

The *rad10* offers the same USB interface as the *HackRF* and can be used with the same driver, which is *hackrf_android*. The driver only needs two minor modifications:

1. If the driver detects a rad10, the option to enable the antenna power is disabled.
2. Because the *rad10* has a different USB product ID, the driver has to be extended in order to also detect a *rad10* as valid *HackRF*.

Both changes have been applied to the *hackrf_android* driver. Unfortunately, the creators of AnSiAn have replaced the driver library with a plain copy of the driver source code, which makes it hard to update it to a newer version. Therefore the driver code was removed from AnSiAn and replaced again with the current version of the driver library.

This is the only change necessary to add support for the new SDR hardware. The *rad10* can now be used by selecting the *HackRF* source in AnSiAn.

### 4.4.2 *SDRPlay*

The *SDRplay* is a low cost SDR with a frequency range between 100 kHz and 2 GHz. It has a 12-bit Analog-to-Digital Converter (ADC) and therefore suffers less from quantification noise than the *HackRF* and the *RTL-SDR* (both having a 8-bit ADC). Additionally it provides passive Radio Frequency (RF) filters on the front end to prevent strong out-of-band noise from distorting the input signal.

Martin Marinov developed an Android driver for the *SDRplay* dongle, which is currently in beta state. The API is similar to the *RTL-SDR* driver which is developed by the same person.

#### 4.4.2.1 *Implementation*

The *SDRplay* driver is started by an Intend with an URL of the format `'iqsrc://...'`. It takes an IP address and TCP port as arguments and will open a socket respectively. The host app can the connect to the socket and will receive the samples via TCP. Control commands such as changing the frequency, sample rate or other parameters can be sent through the TCP socket to the driver. A list of supported commands can be found on the GitHub page of the driver. They are listed in Table 1.

| Command name | Code | Description |
|---|---|---|
| SET_FREQ | 0x01 | Change the tuning (center) frequency of the dongle |
| SET_SAMPLE_RATE | 0x02 | Change the sample rate of the dongle |
| SET_GAIN_MODE | 0x03 | Change gain mode (auto or manual) |
| SET_GAIN | 0x04 | Change gain value (if gain mode is manual) |
| SET_FREQ_CORRECTION | 0x05 | Change frequency correction value (in ppm) |
| SET_IF_TUNER_GAIN | 0x06 | Change Intermediate Frequency (IF) gain |
| SET_TEST_MODE | 0x07 | Turn on test mode |
| SET_AGC_MODE | 0x08 | Activate or deactivate Automatic Gain Control (AGC) |
| SET_DIRECT_SAMPLING | 0x09 | Activate or deactivate direct sampling mode |
| SET_OFFSET_TUNING | 0x0a | Activate or deactivate offset tuning |
| EXIT | 0x7e | Cause the driver to turn off itself |
| GAIN_BY_PERCENTAGE | 0x7f | Change gain value as percentage |
| ENABLE_16_BIT_SIGNED | 0x80 | Enable 16 bit unsigned sample size (SDRplay only) |

Table 1: Commands for the RTL-SDR / SDRplay driver by Martin Marinov

#### 4.4.2.2 *Open Issues*

As the *SDRplay* Android driver is still in beta stage, the support in AnSiAn has also still beta status. 12-bit samples are currently not working correctly and therefore AnSiAn uses the RTL-SDR compatibility mode of the driver which delivers stripped 8-bit samples. However, the 12-bit converter code is ready to be used and might be enabled in future versions of AnSiAn.

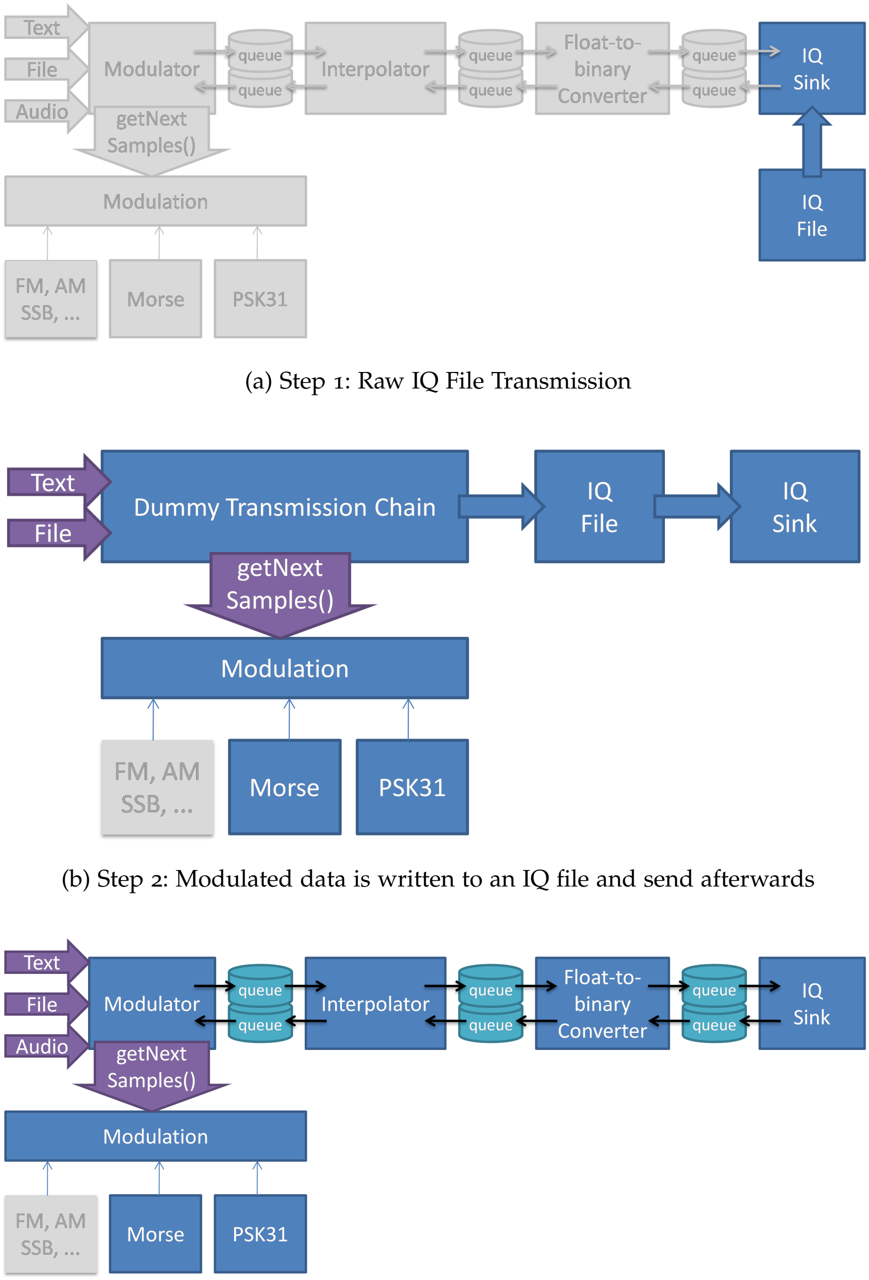
## 4.5 TRANSMISSION

RF Analyzer as well as AnSiAn both have a receive-only signal processing chain, even though they support SDR devices which are capable of transmitting signals, such as the *HackRF*. The implementation of a complete transmission chain is split into multiple steps to go along with the Agile approach (see Figure 14).

In the first step the transmission chain will only include the `IQ Sink` which is able to replay a recorded IQ file. The file has to have the correct format according to the SDR hardware and the sample rate cannot be adjusted.

The second step will provide `Modulators` for some digital and analog modes along with a dummy transmission chain which will use the modulator to create an IQ file and feed it to the `IQ sink` directly.

In the final step the dummy chain is replaced by the complete transmission chain which includes:



(a) Step 1: Raw IQ File Transmission

(b) Step 2: Modulated data is written to an IQ file and send afterwards

(c) Step 3: Real-time modulation, interpolation, conversion and transmission

Figure 14: Incremental implementation steps of the transmission chain

- the `Modulators` from step two
- the `Interpolator` which is able to adjust the sample rate at the output of the `Modulators` to a value that is supported by the `IQ Sink`
- a `float-to-binary` converter which will output the samples in the correct binary format according to the SDR hardware

After the final step the transmission chain will be able to modulate and transmit signals in real-time, assuming the Android device is capable of the necessary calculations.

Step one and two are part of the third sprint and provide a proof-of-concept implementation to demonstrate the transmission capabilities of the app. Step three will be left for future work on the application as well as the addition of more modulators.

### 4.5.1 *Transmission of Raw I/Q Files*

### 4.5.2 *Modulators*

Each `Modulator` will provide an interface to retrieve the next packet of modulated samples. The payload data (e.g. audio or text) might be given at instantiation time or through a queue (in order to enable real-time transmission).

As mentioned above, the step two implementation will include a dummy transmission chain that reads the output of the `Modulator`, converts it to binary IQ data and writes it to a temporary file which will then be transmitted by the `Transmitter` implemented in the previous section.

#### 4.5.2.1 *Morse Modulator*

The morse modulator takes a string as input payload and produces a baseband signal with dits and dahs respectively. The morse encoder from the previous AnSiAn version can be reused to transform the payload text into a sequence of dits ('.'), dahs ('-'), breaks ('␣') and word boundaries ('/'). Each character in this sequence corresponds to a predefined packet of IQ samples (see Table 2).

Because the dit and dahs each also contains silence of the length of one dit at the beginning, the letter separator ('␣') and the word separator ('/') need to be one dit length shorter than they actually are.

To increase the modulation performance, the complex sinusoid is generated ahead of time and the character translation only consists of a copy operation.

| Morse element | Corresponding IQ sample structure |
|---|---|
| . | one dit length of silence + one dit length of tone |
| - | one dit length of silence + three dit lengths of tone |
| ␣ | two dit lengths of silence |
| / | six dit lengths of silence |

Table 2: Translation from morse elements to IQ samples

| Previous state (phase) | Current bit | BPSK output |
|---|---|---|
| 0° | 0 | $\cos(\frac{\pi \cdot 31.25\text{Hz} \cdot t}{f_s})$ |
| 180° | 0 | $-\cos(\frac{\pi \cdot 31.25\text{Hz} \cdot t}{f_s})$ |
| 0° | 1 | 1 |
| 180° | 1 | -1 |

Table 3: Output of the BPSK modulation based on its previous state

#### 4.5.2.2 *PSK31 Modulator*

The PSK31 modulator also takes a string as input payload, encodes it with Varicode and modulates the bits with BPSK. The Varicode dictionary of the PSK31 demodulator can be reused to perform the first step of creating the bitstream from the payload text. The BPSK modulation simply outputs the samples based on the current state (phase) of the modulator (see Table 3).

# 5 CONCLUSION

AnSiAn as fork of RF Analyzer has made huge structural changes to the application architecture in order to implement a clean MVC pattern. Although this has improved the code quality and maintainability it also introduced issues as described in Section 4.1. These issues and their impact on other parts of the app (see Section 4.2.2) had to be addressed before implementing the new features listed in Section 2.1. There were also additional features such as support for the SDRplay hardware or PSK31 transmission, that were added after the initial feature list was created and therefore influenced the time schedule.

Nevertheless, all mandatory features apart from some minor exceptions were successfully implemented. This includes new digital demodulation modes (PSK31 and RDS), logging of demodulator outputs, support for the rad10 badge and transmission functionality along with some basic transmission modes (Morse and PSK31) as well as plain IQ file replay. Not implemented was the FM-modulated transmission of audio files and the optional features: Walky-Talky-Mode and packet radio demodulation.

Some open tasks and improvements remain to be done in future work on the application. These include the implementation of the complete transmission chain as outlined in Section 4.5, additional modulation modes and improvements on the BPSK demodulation algorithm by switching to a PLL-based approach.

# APPENDIX A

| Character | Encoding | Character | Encoding |
|---:|---|---:|---|
| NUL | 1010101011 | ! | 111111111 |
| SOH | 1011011011 | " | 101011111 |
| STX | 1011101101 | # | 111110101 |
| ETX | 1101110111 | $ | 111011011 |
| EOT | 1011101011 | % | 1011010101 |
| ENQ | 1101011111 | & | 1010111011 |
| ACK | 1011101111 | ′ | 101111111 |
| BEL | 1011111101 | ( | 11111011 |
| BS | 1011111111 | ) | 11110111 |
| HT | 11101111 | * | 101101111 |
| LF | 11101 | + | 111011111 |
| VT | 1101101111 | , | 1110101 |
| FF | 1011011101 | - | 110101 |
| CR | 11111 | . | 1010111 |
| SO | 1101110101 | / | 110101111 |
| SI | 1110101011 | 0 | 10110111 |
| DLE | 1011110111 | 1 | 10111101 |
| DC1 | 1011110101 | 2 | 11101101 |
| DC2 | 1110101101 | 3 | 11111111 |
| DC3 | 1110101111 | 4 | 101110111 |
| DC4 | 1101011011 | 5 | 101011011 |
| NAK | 1101101011 | 6 | 101101011 |
| SYN | 1101101101 | 7 | 110101101 |
| ETB | 1101010111 | 8 | 110101011 |
| CAN | 1101111011 | 9 | 110110111 |
| EM | 1101111101 | : | 11110101 |
| SUB | 1110110111 | ; | 110111101 |
| ESC | 1101010101 | < | 111101101 |
| FS | 1101011101 | = | 1010101 |
| GS | 1110111011 | > | 111010111 |
| RS | 1011111011 | ? | 1010101111 |
| US | 1101111111 | @ | 1010111101 |

| | | | |
|---|---|---|---|
| SP | 1 | A | 1111101 |
| B | 11101011 | c | 101111 |
| C | 10101101 | d | 101101 |
| D | 10110101 | e | 11 |
| E | 1110111 | f | 111101 |
| F | 11011011 | g | 1011011 |
| G | 11111101 | h | 101011 |
| H | 101010101 | i | 1101 |
| I | 1111111 | j | 111101011 |
| J | 111111101 | k | 10111111 |
| K | 101111101 | l | 11011 |
| L | 11010111 | m | 111011 |
| M | 10111011 | n | 1111 |
| N | 11011101 | o | 111 |
| O | 10101011 | p | 111111 |
| P | 11010101 | q | 110111111 |
| Q | 111011101 | r | 10101 |
| R | 10101111 | s | 10111 |
| S | 1101111 | t | 101 |
| T | 1101101 | u | 110111 |
| U | 101010111 | v | 1111011 |
| V | 110110101 | w | 1101011 |
| W | 101011101 | x | 11011111 |
| X | 101110101 | y | 1011101 |
| Y | 101111011 | z | 111010101 |
| Z | 1010101101 | { | 1010110111 |
| [ | 111110111 | \| | 110111011 |
| \ | 111101111 | } | 1010110101 |
| ] | 111111011 | ~ | 1011010111 |
| ^ | 1010111111 | DEL | 1110110101 |
| _ | 101101101 | a | 1011 |
| ` | 1011011111 | b | 1011111 |

Table 5: Varicode Table

| Group | Group | |
|---|---|---|
| Type | Version | Description |
| 0 | A | Basic tuning and switching information only |
| 0 | B | Basic tuning and switching information only |
| 1 | A | Programme Item Number and slow labelling codes |
| 1 | B | Programme Item Number |
| 2 | A | RadioText only |
| 2 | B | RadioText only |
| 3 | A | Applications Identification for ODA only |
| 3 | B | Open Data Applications |
| 4 | A | Clock-time and date only |
| 4 | B | Open Data Applications |
| 5 | A | Transparent Data Channels |
| 5 | B | Transparent Data Channels |
| 6 | A | In House applications or ODA |
| 6 | B | In House applications or ODA |
| 7 | A | Radio Paging or ODA |
| 7 | B | Open Data Applications |
| 8 | A | Traffic Message Channel or ODA |
| 8 | B | Open Data Applications |
| 9 | A | Emergency Warning System or ODA |
| 9 | B | Open Data Applications |
| 10 | A | Programme Type Name |
| 10 | B | Open Data Applications |
| 11 | A | Open Data Applications |
| 11 | B | Open Data Applications |
| 12 | A | Open Data Applications |
| 12 | B | Open Data Applications |
| 13 | A | Enhanced Radio Paging or ODA |
| 13 | B | Open Data Applications |
| 14 | A | Enhanced Other Networks information only |
| 14 | B | Enhanced Other Networks information only |
| 15 | A | Defined in RBDS [15] only |
| 15 | B | Fast switching information only |

Table 4: RDS Group Types

# BIBLIOGRAPHY

[1] RDS Forum Office. "The new RDS IEC 62106:1999 standard." In: (1999).

[2] *demantz/RFAnalyzer: Spectrum Analyzer for Android using the HackRF*. `https://github.com/demantz/RFAnalyzer`.

# ERKLÄRUNG

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*Darmstadt, 1. September 2016*

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