RF Testing Applications for Software-Defined Radio

Using SDR for Impairment Generation

A Technical Paper prepared for SCTE•ISBE by

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1. Introduction

Recent advances in digital communications algorithms and artificial intelligence, coupled with the exponential growth of personal computing power, have positioned software-defined radio (SDR) technology as a viable tool for RF signal capture and playback, simulation, and testing. Popular applications for modern SDR designs include:

- Cellular handsets with programmable digital signal processing (DSP)
- University research / teaching aids
- Military and satellite communications platforms that make use of programmable cores for intermediate frequency (IF) and baseband signal processing
- Cognitive radio – a "smart" radio mesh that can adapt to interference adjusting frequency or power for example
- Amateur radio

The goal of this paper is to outline the general architecture and capabilities of a software-defined radio, followed by some basic principles of digital signal processing. A familiarity with sampling theory and the discrete Fourier transform, for example, will be very helpful for anyone looking to get started with SDR.

Finally, two lab applications of software-defined radio will be detailed. Both of these applications were used in testing a profile management application implementation, so a section on PMA is introduced. After familiarization with PMA, the first test example uses SDR to capture long term evolution (LTE) signals, then re-play them back into the RF plant to be subsequently detected by pattern recognition software that is a function of PMA. The second case recreates plant conditions from a cable modem's perspective by translating its reported receive modulation error ratio (RxMER) values into a waveform to be used as an impairment profile applied to an orthogonal frequency division multiplexing (OFDM) channel. Again, this impairment can then be analyzed by the analytics engine of PMA, and a custom OFDM profile recommendation can be verified.

2. Software-Defined Radio General Architecture

A software-defined radio is a radio that replaces traditional hardware components such as mixers, filters, and modulator/demodulators with software implementations, typically running on personal computers and driving a minimalistic hardware set. The Wireless Innovation Forum and IEEE P1900.1 working group has further simplified the definition down to:

"Radio in which some or all of the physical layer functions are software defined" (1)

There are many advantages to software-defined radio designs. Potentially first among these is the low cost. Devices can range from $10 USB-powered dongles, up to $10,000 high performance units with multiple transmit/receive (TX/RX) channels, higher maximum sampling rates, and greater usable bandwidth. Another primary advantage is flexibility. SDRs are not hardware limited to any particular specification standard, modulation type, or operating frequency range, for example. The ability to test new designs in software also aids in the rapid prototyping of communication systems.

Conceptually, an ideal software-defined radio would eliminate any analog signal processing stages, aside from an antenna, power amplifier, and input/output (I/O) sink or source such as a speaker or microphone. In reality, today's SDRs use an analog frequency conversion stage at the receiver, with all signal processing thereafter being digital. Similarly at the transmitter, the digital-to-analog conversion stage is followed by local oscillator/mixer for transmit frequency conversion along with power amplification.

To illustrate a typical SDR design, the block diagram in Figure 1 outlines the functional components. This was the hardware used in the two practical example sections to follow.

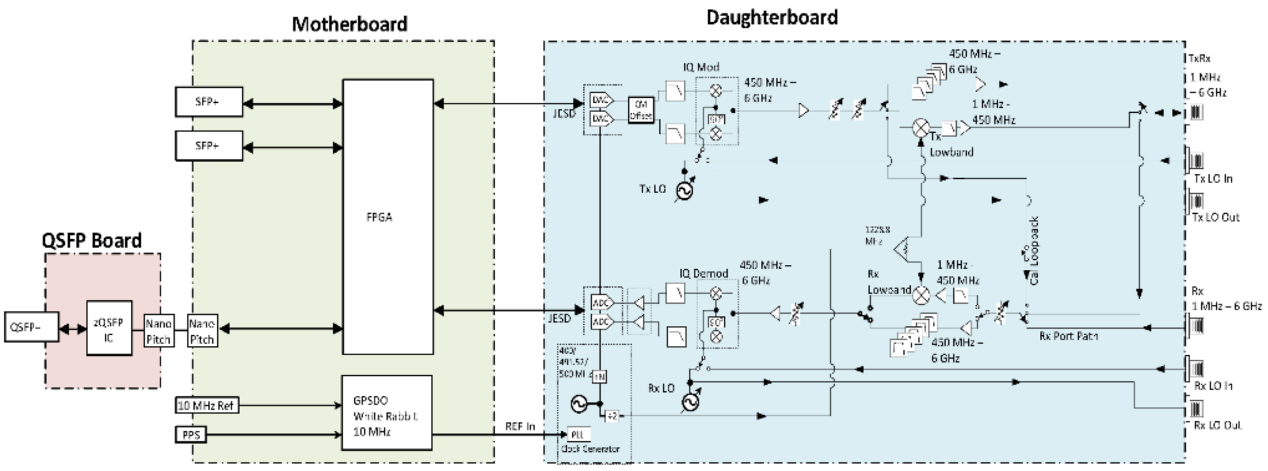


Figure 1 – SDR Architecture Block Diagram

The key features of this design include:

- Xilinx Zynq-7100 FPGA SoC
- Dual-core ARM A9 800 MHz CPU
- Two RX, two TX in half-wide RU form factor
- 3 MHz to 6 GHz frequency range
- Up to 200 MHz of instantaneous bandwidth per channel
- Sub-octave RX, TX filter bank
- 14 bit ADC, 16 bit DAC
- Configurable sample rates: 200, 245.76, 250 MS/s
- Two SFP+ ports (1 GbE, 10 GbE, Aurora, White Rabbit)
- One QSFP+ port (2x 10 Gb / Aurora)
- RJ45 (1 GbE)
- 10 MHz clock reference
- PPS time reference

- External RX, TX LO input ports
- Built-in GPSDO

One of the main factors to consider in selecting a radio is usable bandwidth. The specification above lists 200 MHz of available bandwidth that will allow us to simulate or impair a full OFDM block with 190 MHz of modulated spectrum. Another point to consider is sampling rate. Radios range from fixed, to selectable, to fully configurable sampling rates, with the fully configurable ones being the most flexible, because in some cases they can eliminate the need for additional resampling functions. Many designs use open-source Verilog code for the FPGA that can be modified for high performance applications.

3. SDR Software: GNU Radio

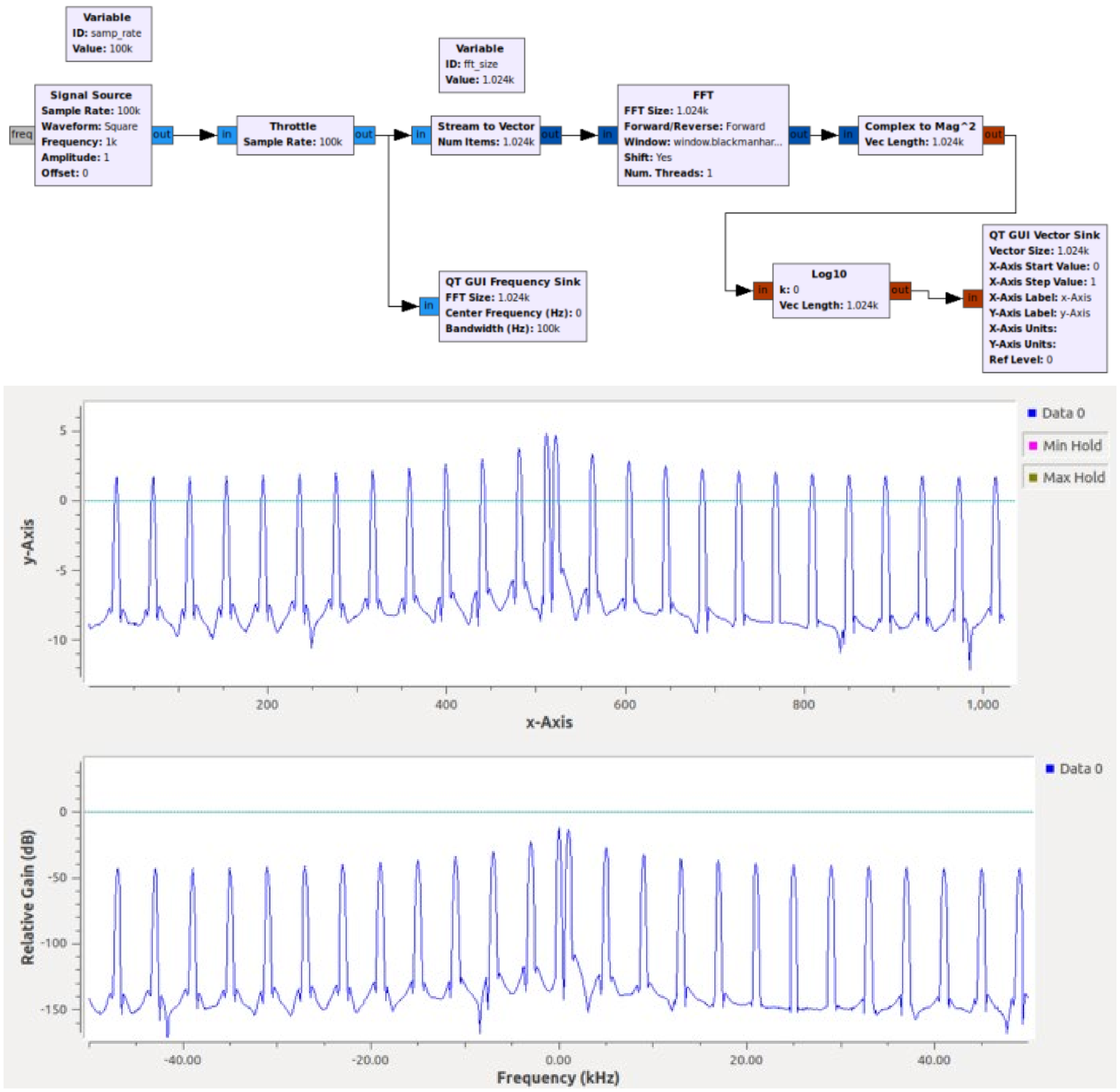
GNU Radio is a free open-source development project that implements signal processing blocks in software for waveform creation, sampling, and analysis, that supports a variety of software-defined radio hardware. It is used extensively in academic and commercial environments for wireless communications research and real-world radio systems.

GNU Radio consists of software-based signal processing blocks used in conjunction with hardware drivers to build applications capable of transmitting or receiving digital data streams. These streams can be directed to disk and saved in different file formats, or to a software-defined radio sink. GNU Radio resource blocks include equalizers, filters, modulators/demodulators, filters, and other common radio elements. These blocks' inputs and outputs are then connected in software to build an end-to-end flow.

Being software based, GNU Radio operates on digital data. It uses complex baseband samples as input for receivers and as the output data type for transmitters. Analog mixing hardware can then shift the signal to the proper frequency.

One of the biggest advantages to working with GNU Radio is that it is primarily written using the Python programming language, with the computationally intensive operations written in C++. GNU Radio also incorporates a graphical user interface – creatively named GNU Radio Companion – that offers drag-and-drop software blocks to construct flowgraph applications. These GRC flowgraphs present an intuitive approach to thinking about the discrete signal processing stages. Once a flowgraph contains a source and sink, the file can be saved and executed, and Python code is automatically generated. At this point there is no need to run the flow from within GNU Radio Companion, as the Python code can be executed directly from the command line or Python shell.

The GRC flowgraph shown in Figure 2 is an example of how a fast Fourier transform (FFT) block can be used to reproduce the same output as another GRC provided block: the Qt GUI frequency sink. The only difference is that Qt GUI frequency sink has the FFT, stream-to-vector, magnitude, and log functions built in.



Variable
ID: samp_rate
Value: 100k

Variable
ID: fft_size
Value: 1.024k

Signal Source
Sample Rate: 100k
Waveform: Square
Frequency: 1k
Amplitude: 1
Offset: 0

Throttle
Sample Rate: 100k

Stream to Vector
Num Items: 1.024k

FFT
FFT Size: 1.024k
Forward/Reverse: Forward
Window: window.blackmanhar...
Shift: Yes
Num. Threads: 1

Complex to Mag^2
Vec Length: 1.024k

QT GUI Frequency Sink
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 100k

Log10
k: 0
Vec Length: 1.024k

QT GUI Vector Sink
Vector Size: 1.024k
X-Axis Start Value: 0
X-Axis Step Value: 1
X-Axis Label: x-Axis
Y-Axis Label: y-Axis
X-Axis Units:
Y-Axis Units:
Ref Level: 0

Figure 2 - GNU Radio Flowgraph Example

In my experience, it is best to compile GNU Radio, SDR drivers, and associated software from source code for your specific operating system. GNU Radio can compile on Windows, Mac OS, and Linux. Ubuntu 20.04 has been used for the testing contained herein.

4. Sampling Theory

When beginning to work with software-defined radio, it may be helpful to review some basic principles of signal processing.

In digital signal processing, sampling of a continuous-time input signal is performed in an analog-to-digital converter (ADC). The output of the converter is a series of data points that sample the instantaneous value of the input signal, and are taken at a given sampling rate. These discrete samples may be used to perform filtering or other processing in the digital domain using a digital signal processor, and then converted back to the analog domain using a digital to analog converter (DAC) as needed. Key to the sampling process is the sampling rate, or the number of samples taken per second (measured in Hz).

Different sampling rates are specified for different purposes. For example, 44.1 kHz is the standard sampling rate used in CDs, and 48 kHz is the recommended sampling rate for many other applications, including audio in digital video recordings. If the signal is sampled properly, the original analog signal may be reconstructed perfectly from the digital samples.

In order to choose the sampling rate for a specific application, the concept of aliasing is important to consider. Aliasing refers to the phenomenon that results in an imperfectly reconstructed analog signal from digital samples created when the sampling rate is too low. As we can see in the illustration in Figure 3, the blue signal has been reconstructed erroneously from the red signal due to the inadequate number of samples taken.

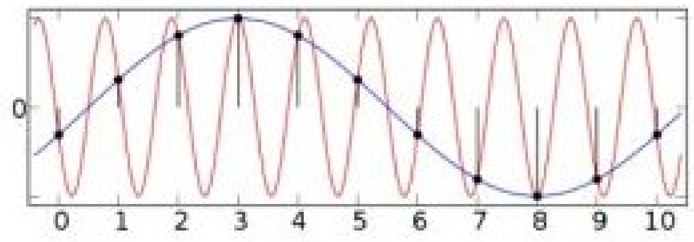


Figure 3 – Aliasing Due to Undersampling

In order to avoid aliasing and accurately reconstruct a signal from its samples, the Nyquist condition must be satisfied. This condition requires that the sampling frequency f_s must be greater than or equal to two times the frequency bandwidth B of the input signal $f(t)$:
$N = f_s \geq 2B$

As long as the Nyquist criterion is met, it is theoretically possible to exactly reconstruct $f(t)$ from its discrete samples f_n, where

$$f_n \equiv f(nT), -\infty < n < \infty$$

The samples of f_n are spaced at time $t = nT$, where T is the sampling interval $\frac{1}{f_s}$.

Under the Nyquist sampling condition, $f(t)$ can be reconstructed from its discrete samples f_n by using the reconstruction formula:

$$f(t) = \sum_n f_n sinc(\frac{\pi}{T}(t - nT))$$

The sinc function, $sinc(t) = \frac{sin(t)}{t}$ is also known as the interpolating function.

5. The Discrete Fourier Transform or DFT

Mathematically, the Fourier series is used to represent any periodic function as a sum of sine and cosine functions. If we have any $f(t)$ with period $T = \frac{2\pi}{\omega_0}$, then the Fourier series is

$$f(t) = \sum_{n=\infty}^{\infty} F_n e^{jn\omega_0 t}$$

with the Fourier coefficients

$$F_n = \frac{1}{T} \int_T f(t) e^{-jn\omega_0 t} dt$$

where Euler's formula is used to give us our sinusoids:

$$e^{jn\omega_0 t} = cos(n\omega_0 t) + j sin(n\omega_0 t)$$

where ω_0 represents frequency, t represents time, and j represents the imaginary number $j = \sqrt{-1}$.

To visualize the construction of a periodic function as a sum of sines, the animation in Figure 4 depicts the construction of a square wave that becomes closer to the ideal with higher frequency resolution:

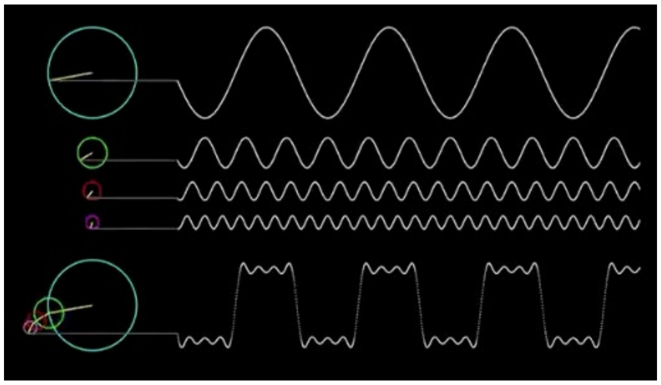


Figure 4 – Square Wave Construction From Sine Waves

If a function is aperiodic, it can still be represented as a sum of sinusoids, but the frequencies of the sinusoids are no longer simply harmonics of a fundamental ω_0. A general aperiodic signal consists of a continuous spectrum of frequencies, so rather than integrating over a single period of $f(t)$ as is the case with Fourier series coefficients, we must integrate $f(t)$ over all time to derive the exact Fourier transform, which is the equivalent to the Fourier series coefficients but in a continuous frequency range. In the limit as the period $T = \frac{2\pi}{\omega_0}$ approaches infinity in an aperiodic signal, then the fundamental frequency ω_0 approaches 0, and so we must integrate over the entire period of an aperiodic signal, which in other words, means we integrate over all time:

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}dt, \omega \in (-\infty, \infty)$$

In analog signal processing, the Fourier transform $F(\omega)$ is used to determine the characteristics of the signal $f(t)$ in the frequency domain. We can determine the frequencies or spectral content present in $f(t)$ using the Fourier transform, and manipulate or filter the signal as needed in the frequency domain.

Then, we can use an inverse Fourier transform to go back to the original time domain function from the frequency domain:

$$f(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} F(\omega)e^{j\omega t}d\omega$$

Similarly, in digital signal processing, the discrete-time Fourier transform (DTFT) is used in order to convert a continuous discrete-time signal (input sequence) into its counterpart in the continuous frequency domain. It is modeled by the equation

$$X(\omega) = \sum_{n=-\infty}^{\infty} x[n]e^{-j\omega n}, \omega \in [-\pi, \pi)$$

where $x[n]$ represents a discrete-time input signal, and n is an integer.

The inverse DTFT is given by

$$x[n] = \frac{1}{2\pi}\int_{-\pi}^{\pi} X(\omega)e^{j\omega n}d\omega, \omega \in [-\pi, \pi)$$

Finally, the discrete Fourier transform (DFT) is simply a "sampling" of the DTFT in the frequency domain – a finite version of the DTFT taken over a specified interval or number of samples. It is modeled by the equation

$$X[k] = \sum_{n=0}^{N-1} x[n]e^{\frac{-j2\pi kn}{N}}, k = 0, 1, \ldots, N-1$$

where N represents the total number of samples.

And we can also define the inverse DFT by:

$$x[n] = \frac{1}{N}\sum_{k=0}^{N-1} X[k]e^{\frac{j2\pi kn}{N}}, n = 0, 1, \ldots, N-1$$

The DFT is a mainstay in the digital signal processing toolbox. It allows us to determine the spectral components of a sampled signal and to determine the frequency response of a system given the impulse response of the system in a completely digital manner. It uses a finite number of samples which can be easily used in computations of all kinds.

The FFT, or fast Fourier transform, is an algorithm that is used to calculate the DFT in a mathematically efficient manner. The FFT is employed by mathematicians, engineers, musicians, and scientists to rapidly analyze, modify, or utilize sampled signals in the frequency domain for myriad applications.

6. Introduction to Profile Management Application

The next two sections of this paper provide examples of using SDR to generate impairments for profile management application testing. A brief overview of PMA will demonstrate how highly customizable impairment profiles can aid in OFDM profile management testing.

A profile management application system builds an optimized set of OFDM channel modulation profiles based on current plant conditions. It does this by collecting receive modulation error ratio data from cable modems, analyzing this data to sort modems into impairment groups, and finally building the modulation profiles and applying them on the CMTS (see Figure 5). The modulation profiles themselves typically contain variable bit-loaded regions ranging from excluded regions (0-QAM) for highly impaired subcarriers, to 4096-QAM in the low noise areas of the OFDM channel. In this way, the PMA algorithm is able to maximize overall channel capacity while maintaining robust profiles that are noise-tolerant.

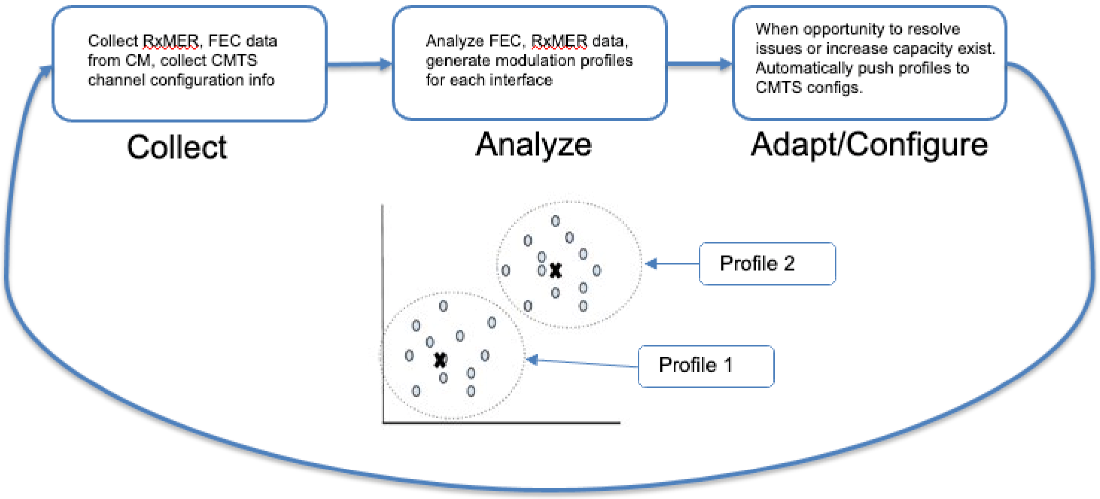

Collect RxMER, FEC data from CM, collect CMTS channel configuration info
Collect
Analyze FEC, RxMER data, generate modulation profiles for each interface
Analyze
When opportunity to resolve issues or increase capacity exist. Automatically push profiles to CMTS configs.
Adapt/Configure
Profile 2
Profile 1

Figure 5 – The Profile Management Application System

Laboratory RF environments are often very clean, with short cable runs, limited ingress, and few taps or active elements such as amplifiers or line extenders. To test profile recommendations generated by the PMA algorithm (Figure 6) we will need to introduce impairments with variable power levels, widths, and durations. The following sections demonstrate how software-defined radio can be used to create these impairments.

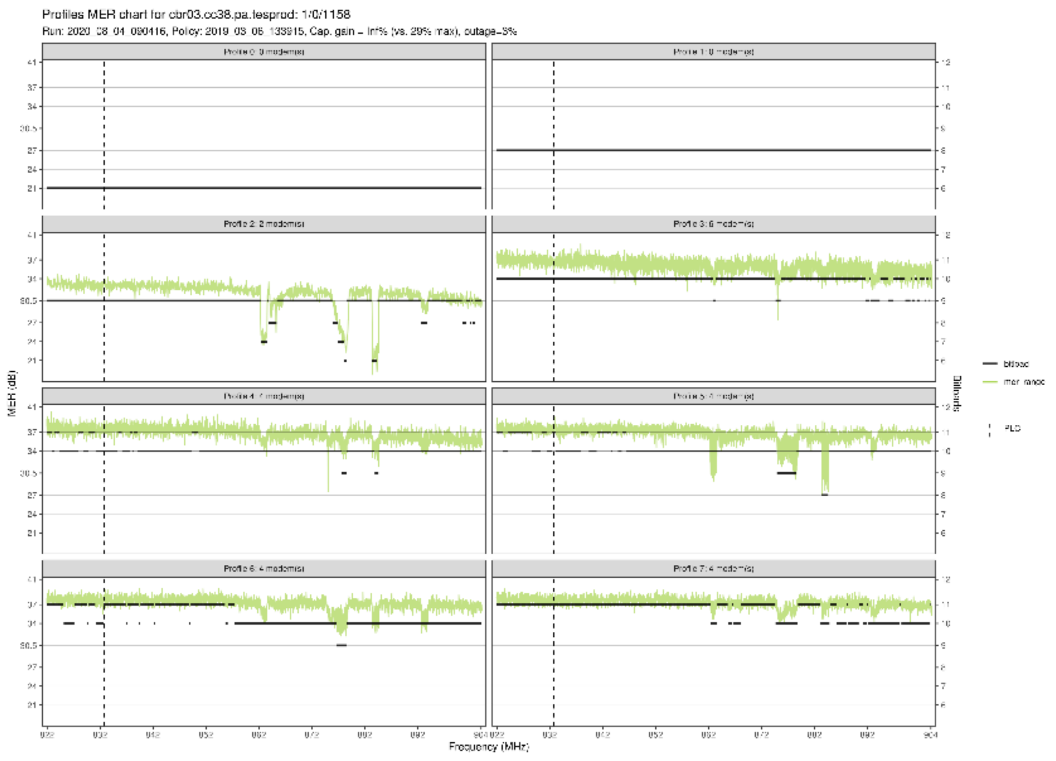


Figure 6 – PMA In Action: Building Custom OFDM Profiles

7. Practical Application Example #1– Capturing LTE Signals For Replay as an Impairment Source

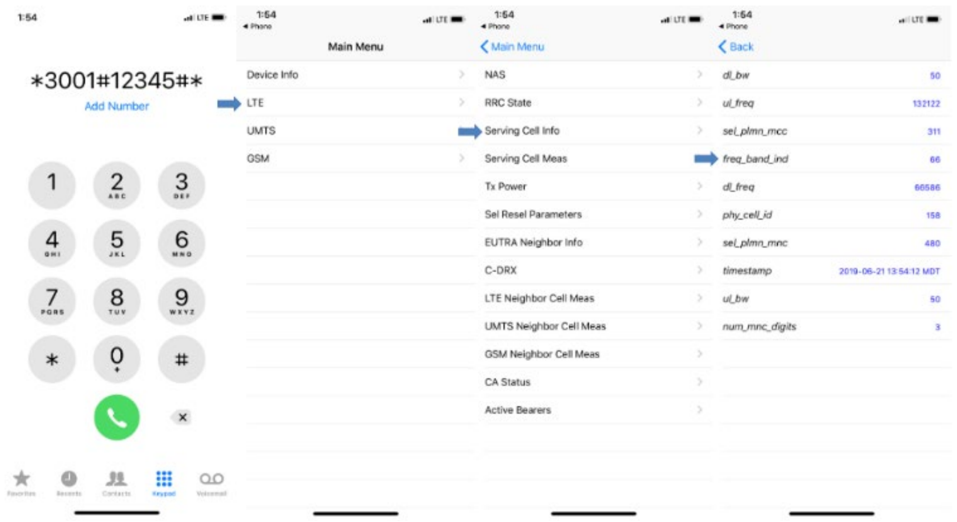
Problem Statement: How can we capture LTE signals and introduce them as ingress interference in a laboratory environment?

SDR Solution Outline:

1. Using a cellular telephone as a signal source, find the LTE band currently in use
2. Prepare the SDR to capture at the LTE uplink frequency
3. Start a speed test on the UE to generate traffic
4. Observe/capture the LTE signal on the SDR
5. Combine SDR TX into the lab RF plant and playback capture file
6. Observe effect in the cable modem's downstream OFDM channel and verify PMA OFDM channel profile recommendation

Detailed Steps:

1. Most modern UE devices include hidden service menu interfaces that can be accessed by dialing certain sequences. For example:
    a. Apple iPhone (all): *3001#12345#*
    b. Samsung (Android): *#*#197328640#*#*or *#0011#
    c. Sony (Android): *#*#*386#*#* or *#*#*585*0000#*#*
    d. HTC (Android): *#*#7262626#*#*



dl_bw	50
ul_freq	132122
sel_plmn_mcc	311
freq_band_ind	66
dl_freq	66586
phy_cell_id	158
sel_plmn_mnc	480
timestamp	2019-06-21 13:54:12 MDT
ul_bw	50
num_mnc_digits	3

Figure 7 - Finding LTE Band In Use on iPhone

From the above example, we can determine what uplink frequencies are assigned to Band 66:

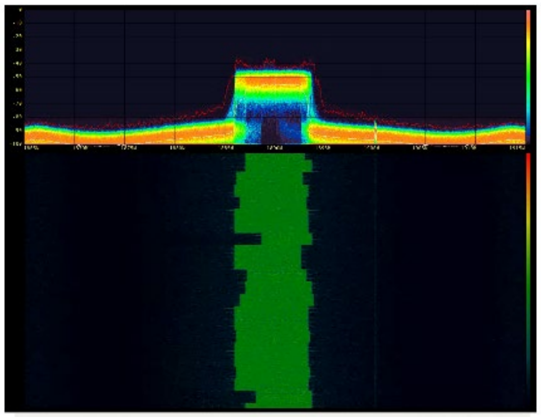
Table 1 – LTE Frequency Bands (3)

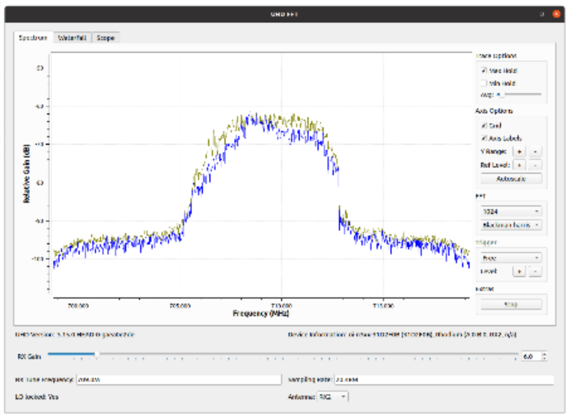
Band	Duplex mode[A 1]	f (MHz)	Common name	Subset of band	Uplink[A 2] (MHz)	Downlink[A 3] (MHz)	Duplex spacing (MHz)	Channel bandwidths (MHz)
2	FDD	1900	PCS[A 4]	25	1850 – 1910	1930 – 1990	80	1.4, 3, 5, 10, 15, 20
13	FDD	700	Upper SMH[A 7]		777 – 787	746 – 756	−31	5, 10
66	FDD	1700	Extended AWS (AWS-1–3)[A 17]		1710 – 1780	2110 – 2200[2]	400	1.4, 3, 5, 10, 15, 20

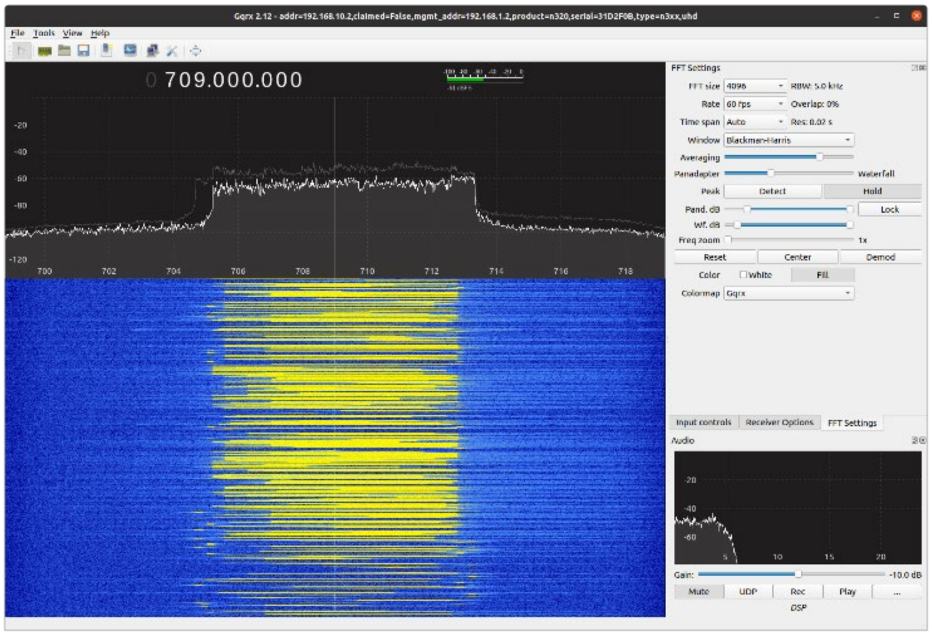
2. Osmocom_fft is a simple spectrum analyzer and capture tool that can be used to write sampled data directly to a file. Using a command such as:
   osmocom_fft -f 1.89e9 -s 40.96e6
   will launch the application and tune to 1.89 GHz.
   If the Osmocom utilities are not available, a GNU Radio Companion flowgraph can be created to output the data from a UHD:USRP source to a file sink block.
   The RX port of the SDR should be fitted with an antenna capable of receiving the LTE frequencies, such as a mini GSM/cellular quad-band antenna with 2 dBi of gain and 50 ohm impedance.

3. Install Speed test on the UE LTE source, disable Wi-Fi, and begin a speed test. Most testing includes a downlink and an uplink speed test portion. If the energy is not seen at the expected frequency, the downlink test may not have completed or the UE may have changed bands or moved to a neighboring macro cell.

4. The LTE signal should now be see on the analyzer. For live, raw IQ capture, Gqrx is a free open-source software-defined radio receiver that supports many popular radios including Airspy, rtl-sdr, HackRF, and USRP devices.






Gqrx 2.12 - addr=192.168.10.2,claimed=False,mgmt_addr=192.168.1.2,product=n320,serial=31D2F0B,type=n3xx,uhd

709.000.000

FFT Settings
FFT size 4096 RBW: 5.0 kHz
Rate 60 fps Overlap: 0%
Time span Auto Res: 0.02 s
Window Blackman-Harris
Freq zoom 1x
Color ☐ White
Colormap Gqrx
Gain: -10.0 dB

Figure 8 - Real Time Spectrum Analyzers for IQ Capture (clockwise): osmocom_fft, uhd_fft, gqrx SDR

5. Combine the SDR TX Port 1 into the laboratory RF plant as needed to impact the desired set of modems. For example:

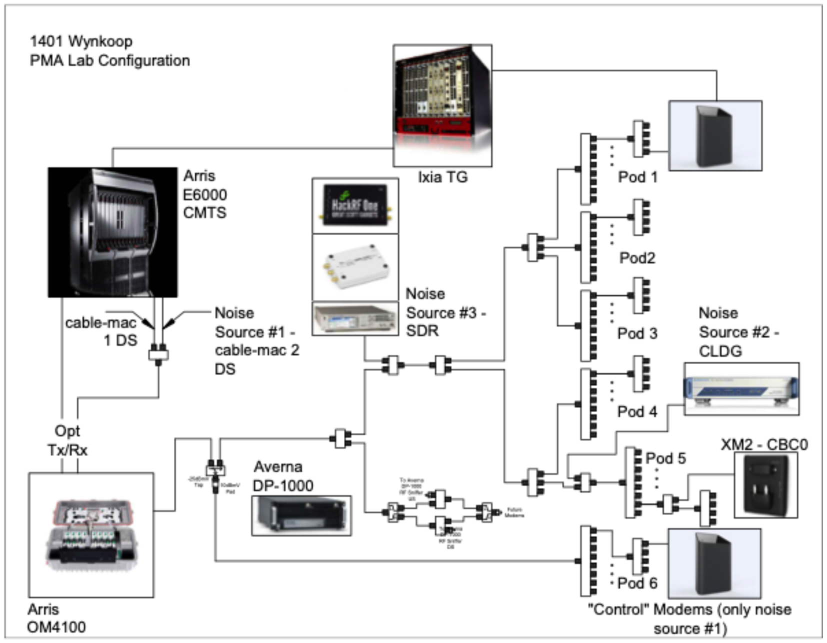


Figure 9 - Combining of Noise Sources in Laboratory Environment

7. Through the splitting and combining network, different interference groups can be created. In this way the k-means based clustering algorithm that the PMA analytics engine implements can be fully tested, as multiple, custom OFDM channel profiles are supported.

8. Practical Application Example #2 – How SDR Has Aided Profile Management Application (PMA) Testing

Problem Statement: How can we re-create an OFDM channel impairment profile using a cable modem's reported RxMER values?

SDR Solution Outline:

1. Read in cable modem RxMER data file
2. Convert the data from ¼ dB values to dB
3. To create the impairment profile, set all values that are within a threshold (say 6-10 dB) of the maximum value to zero. We will be using the values outside of this "good" range to create our impairment.
4. Convert the RxMER value from a ratio of power values to output voltage for the DAC
5. Normalize the voltage value vector
6. Use GNU Radio to stream out the data to an inverse fast Fourier transform block to convert from frequency to time domain
7. Stream the data to either a file to use for later playback, or directly to the software-defined radio sink
8. Inject this RF impairment into the lab environment physical plant to re-create the impairment seen at the customer premises

Detailed Steps:

1. The cable modem RxMER values can be obtained either directly from CMTS CLI or via TFTP download. The values are reported in quarter-dB, for example:

```
0x0000 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0020 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0040 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0060 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
                                          .
                                          .
                                          .
0x0400 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0420 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0440 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0460 00000000 0000B6B7 B9B6BCB6 B5BABAB4 B6B3B4B5 BAB3B8B3 BAB6B8B5 B5BBBBB6
0x0480 B6BAB5B9 B9B7BBB7 BCB4B4BA B4B6B7B9 B6B8B3BD B7B7B8B5 B8B8B8B6 B7B8B7B9
0x04A0 BAB3B8B5 B6B7B8B8 BCB7B8BB B8BAB4B8 B7B6B9B3 B6B9B9B8 B7B9B9B9 B4BCBAB5
0x04C0 B6B8B4B7 B7BAB8B8 B3B8B7B4 B8B6B7B9 B9B4B8B5 B9BAB6B9 B4B6B4B9 B4B6B9BA
0x04E0 B6B9B3B7 BAB2B5B6 B5B5B4B6 B4B7B7BA B9B7B5BA B9B6B9B4 B6B7B7B7 BDBAB6B3
0x0500 B6B6B4B7 B6BABBBC B3B8B5B7 B7B7B5BB B6B6B6B8 B7B7B6BA B4B6B8B9 B7B5B8B7
0x0520 B9B6B5B8 B3B9B7B6 B6BBB7B5 B8B9B4B7 B8B7B4B7 B7B7B8BA B8B7B5B4 B4B4B9B3
                                          .
                                          .
                                          .
0x0F40 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0F60 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
```

```
0x0F80 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0FA0 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0FC0 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
0x0FE0 00000000 00000000 00000000 00000000 00000000 00000000 00000000 00000000
```

If using the values from the TFTP downloaded RxMER file, the RxMER per subcarrier data follows after a header using the specified format:

Table 2 – RxMER TFTP File Format (4)

Element	Size
File type (value = 504E4E04)	4 bytes
Major Version (Value = 1)	1 byte
Minor Version (Value = 0)	1 byte
Capture Time	4 bytes
DS Channel Id	1 byte
CM MAC Address	6 bytes
Subcarrier zero frequency in Hz	4 bytes
FirstActiveSubcarrierIndex	2 bytes
Subcarrier spacing in kHz	1 byte
Length in bytes of RxMER data	4 bytes
Subcarrier RxMER data	RxMerData

The other data such as the subcarrier zero frequency, first active subcarrier index, subcarrier spacing, and length in bytes of the RxMER data will all be used to fully describe the OFDM channel impairment profile.

2. The RxMER data itself is defined as the ratio of the average power of the ideal QAM constellation to the average error-vector power. The error vector is the difference between the equalized received pilot or preamble value and the known correct pilot value or preamble value. The reported values are represented in hexadecimal values of ¼ dB. Multiple each value in the array by ¼ to properly scale.

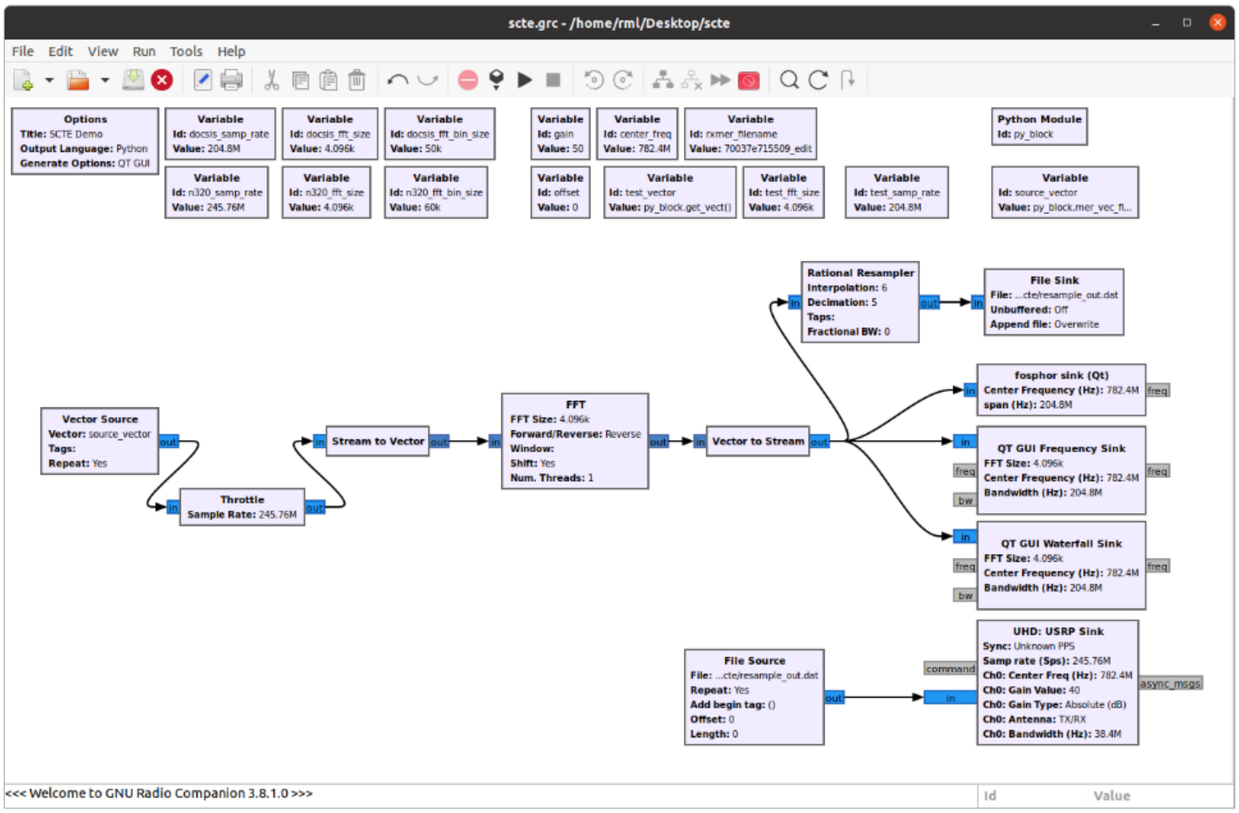
3. In this step we discard (set to zero) all values that are within a configurable threshold of the maximum reported RxMER value. Only the low RxMER values will be used to construct the impairment vector. This example used a value of 10 dB.

4. Convert every value in the impairment array to an output voltage value for the DAC using the formula:

   $V = 10^{(dB/20)}$

5. Divide every voltage value by the sum of values (new values will all sum to 1) to avoid overloading the SDR output stage.

6. The processing of the RxMER data, along with the streaming to either file or directly to the software-define radio sink was achieved through the creation of a GNU Radio Companion flowgraph (Figure 10).


scte.grc - /home/rml/Desktop/scte
File Edit View Run Tools Help

Options
Title: SCTE Demo
Output Language: Python
Generate Options: QT GUI

Variable
Id: docsis_samp_rate
Value: 204.8M

Variable
Id: docsis_fft_size
Value: 4.096k

Variable
Id: docsis_fft_bin_size
Value: 50k

Variable
Id: gain
Value: 50

Variable
Id: center_freq
Value: 782.4M

Variable
Id: rxmer_filename
Value: 70037e715509_edit

Python Module
Id: py_block

Variable
Id: n320_samp_rate
Value: 245.76M

Variable
Id: n320_fft_size
Value: 4.096k

Variable
Id: n320_fft_bin_size
Value: 60k

Variable
Id: offset
Value: 0

Variable
Id: test_vector
Value: py_block.get_vect()

Variable
Id: test_fft_size
Value: 4.096k

Variable
Id: test_samp_rate
Value: 204.8M

Variable
Id: source_vector
Value: py_block.mer_vec_fl...

Vector Source
Vector: source_vector
Tags:
Repeat: Yes

Throttle
Sample Rate: 245.76M

Stream to Vector

FFT
FFT Size: 4.096k
Forward/Reverse: Reverse
Window:
Shift: Yes
Num. Threads: 1

Vector to Stream

Rational Resampler
Interpolation: 6
Decimation: 5
Taps:
Fractional BW: 0

File Sink
File: ...cte/resample_out.dat
Unbuffered: Off
Append file: Overwrite

fosphor sink (Qt)
Center Frequency (Hz): 782.4M
span (Hz): 204.8M

QT GUI Frequency Sink
FFT Size: 4.096k
Center Frequency (Hz): 782.4M
Bandwidth (Hz): 204.8M

QT GUI Waterfall Sink
FFT Size: 4.096k
Center Frequency (Hz): 782.4M
Bandwidth (Hz): 204.8M

File Source
File: ...cte/resample_out.dat
Repeat: Yes
Add begin tag: ()
Offset: 0
Length: 0

UHD: USRP Sink
Sync: Unknown PPS
Samp rate (Sps): 245.76M
Ch0: Center Freq (Hz): 782.4M
Ch0: Gain Value: 40
Ch0: Gain Type: Absolute (dB)
Ch0: Antenna: TX/RX
Ch0: Bandwidth (Hz): 38.4M

```
<<< Welcome to GNU Radio Companion 3.8.1.0 >>>
```

Figure 10 - GRC Flowgraph to Contruct and Transmit Custom Impairment Waveform

<u>Flowgraph Details:</u>
Embedded Python Module-
The previous six steps were all done in the embedded Python Module block. GNU Radio companion allows you to create and use custom Python code within your flowgraph. In this example, all the RxMER data manipulation and plot generation was done within the embedded Python block. After reading in the modem's RxMER data, the values are written to disk as a Matplotlib Pyplot (Figure 11):

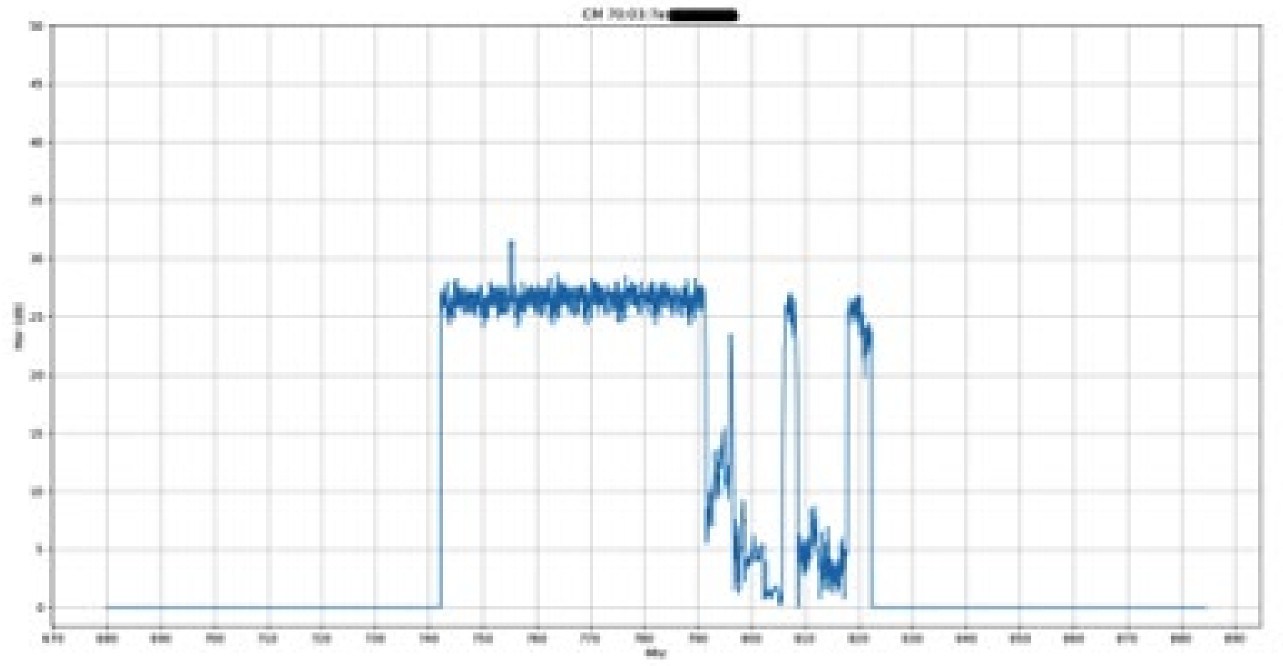


Figure 11 - Graph of RxMER Data Generated by Embedded Python Block

Vector Source Block -
This flowgraph begins with the vector source block. This block streams items based on the input vector, which in this case is the output of a Python method in our embedded Python block.

Throttle Block -
This stream feeds a throttle block. This block will be set to “bypass” when using the real SDR hardware which will set its own sampling rate. It is used in combination with the software sinks or file output, when the sampling rate would only be determined by the host’s CPU clock.

Stream to Vector Block –
This block converts the stream of items into a stream of vectors containing N number of items. In this case the number of items will be the FFT size (4096). The stream-to-vector and vector-to-stream blocks are special in that the input and output types differ by a decimation or interpolation factor. In our example, the stream-to-vector block produces one output vector for every 4096 input samples.

FFT Block –
The complex valued output vector from the previous stage feeds the fast Fourier transform block. The key parameters of the FFT block are: FFT size (number of samples used in each FFT calculation), direction (forward for FFT, reverse for inverse FFT), window, shift (puts DC – 0 Hz – in the center of the FFT block for complex input type), and number of threads.

Vector to Stream Block –
The output of the FFT block will always be complex-valued. This block will convert a stream of vectors from the FFT block to a stream of items that can be fed directly to an output file, the

SDR hardware, a GUI frequency sink that offers spectrum analyzer functionality, or a Fosphor or Waterfall sink for more real-time spectrum analyzer-like output.

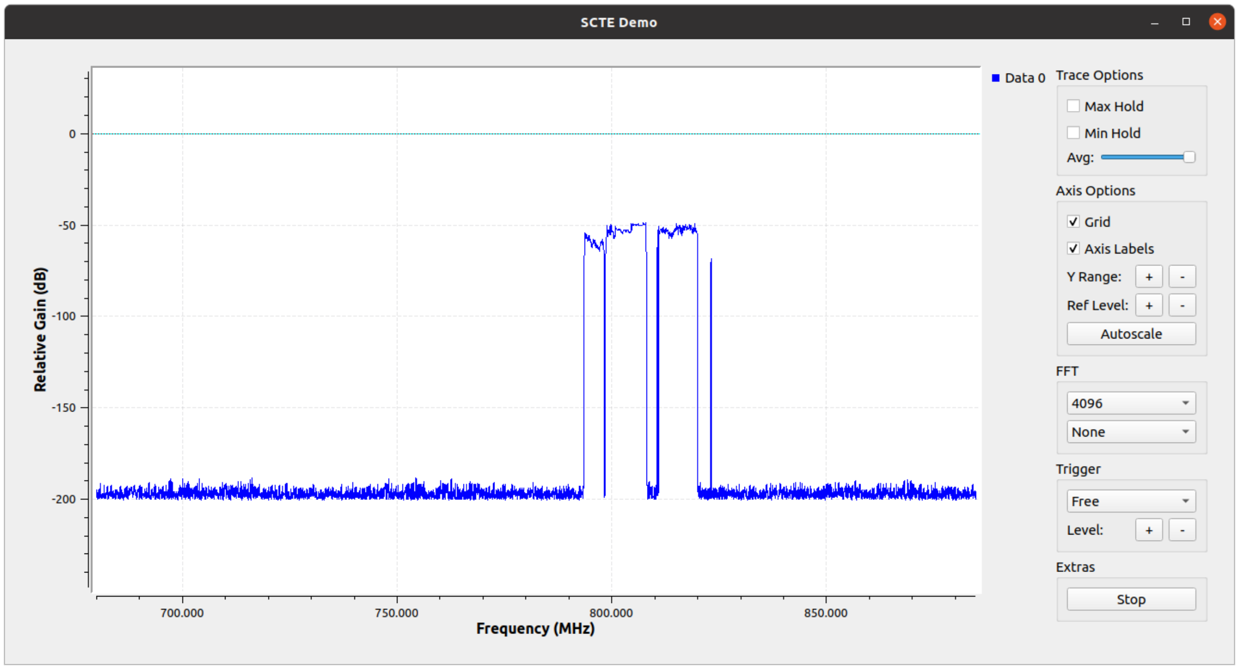


Figure 12 - Qt GUI Frequency Sink Visualization of Impairment Signal

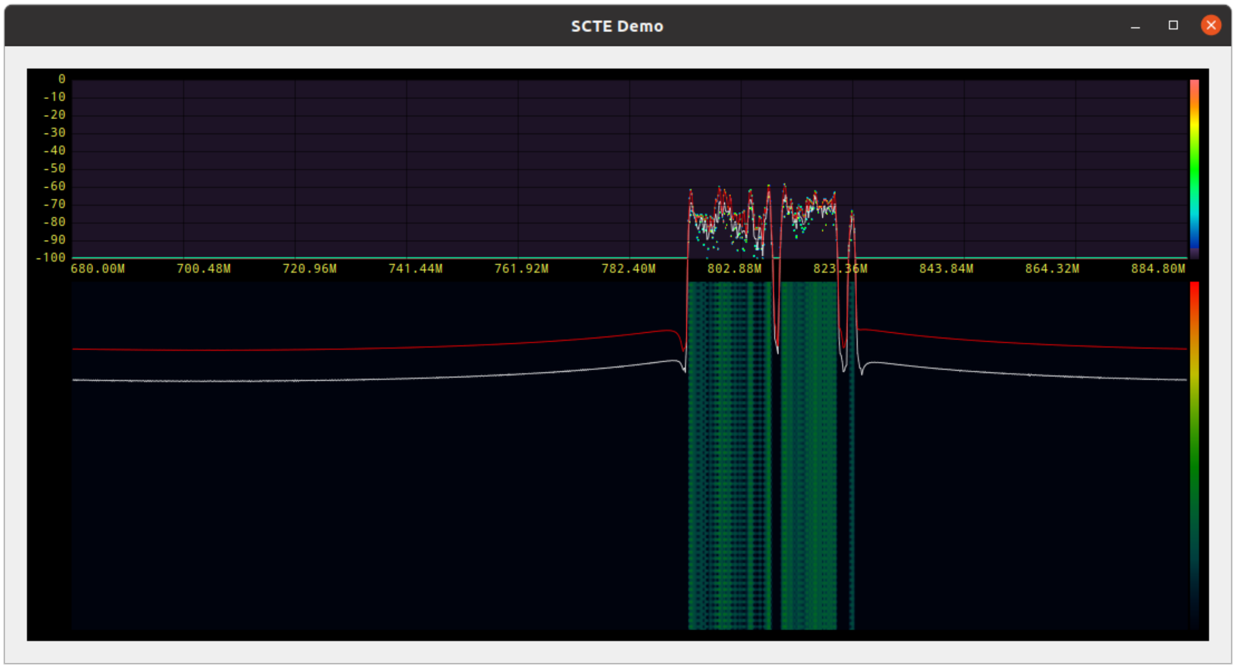
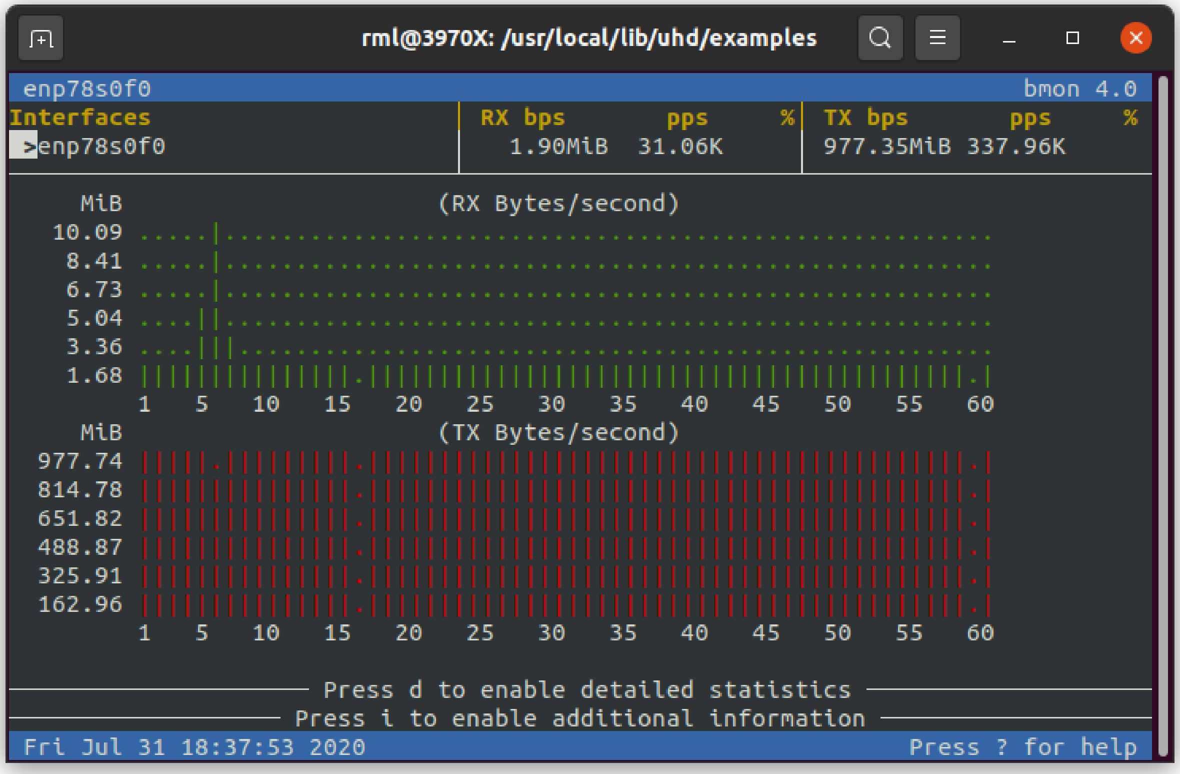


Figure 13 - Qt Fosphor Sink Visualization of Impairment Signal

Rational Resampler Block –
If trying to stream a file that has been sampled at a rate that is not supported by the SDR hardware, the rational resampler block allows us to change the sampling rate (e.g., from the DOCSIS standard 204.8 MHz). In this example the default hardware sampling rate is 245.76 MHz so we could set the interpolation factor to 6 and the decimation factor to 5 to upsample to 245.76 MHz.

7. If the output is sent to a file, we can replay directly to the SDR sink without continuously performing the IFFT calculation repeatedly. The act of streaming 32 bit IQ data at 245.76 MHz is very I/O intensive and requires a powerful host computer with a 10 gigabit interface. As seen in Figure 14 this stream from the host to the SDR hardware averages around ~7.8 Gbps:



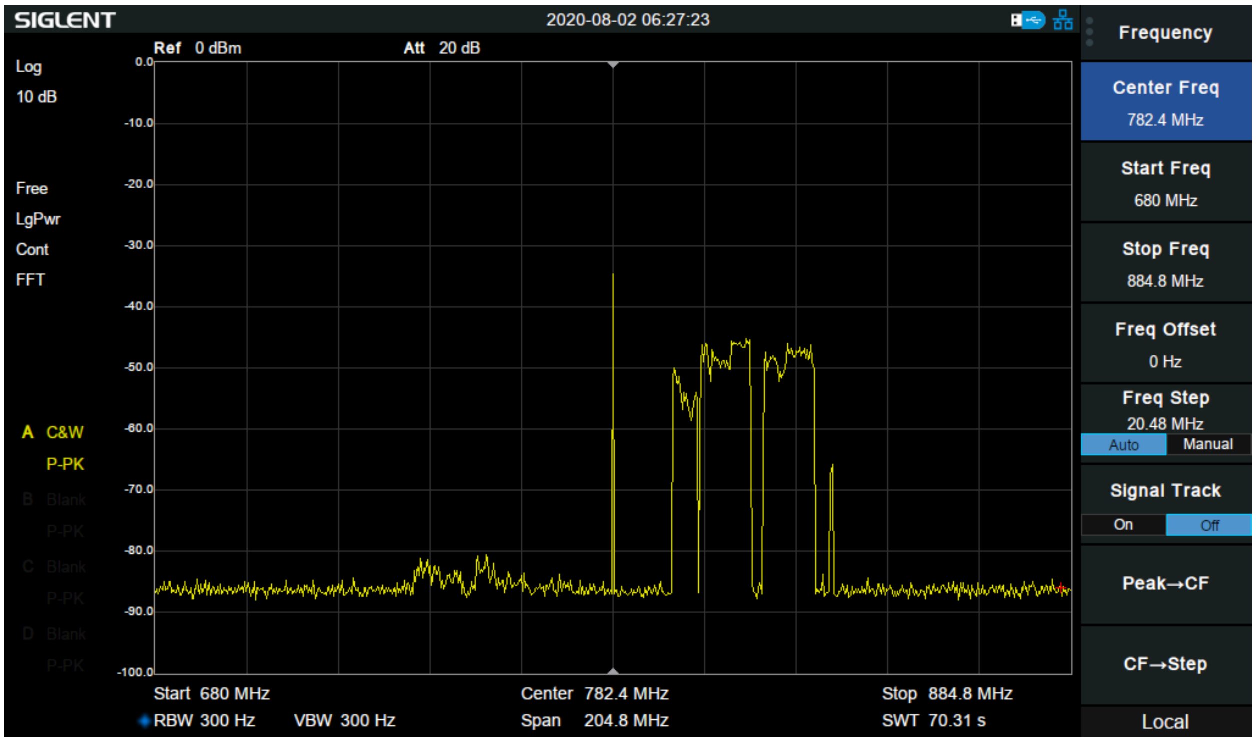
```
rml@3970X: /usr/local/lib/uhd/examples
enp78s0f0                                                                   bmon 4.0
Interfaces                      │ RX bps        pps     %│ TX bps        pps     %
>enp78s0f0                      │   1.90MiB   31.06K     │ 977.35MiB  337.96K

     MiB                     (RX Bytes/second)
   10.09 .....|.....................................................
    8.41 .....|.....................................................
    6.73 .....|.....................................................
    5.04 ....||.....................................................
    3.36 ....|||....................................................
    1.68 ||||||||||||||.|||||||||||||||||||||||||||||||||||||||||||.|
         1   5    10   15   20   25   30   35   40   45   50   55   60
     MiB                     (TX Bytes/second)
  977.74 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
  814.78 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
  651.82 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
  488.87 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
  325.91 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
  162.96 |||||.|||||||||.||||||||||||||||||||||||||||||||||||||||||.|
         1   5    10   15   20   25   30   35   40   45   50   55   60

─────────────────── Press d to enable detailed statistics ──────────────────
────────────────── Press i to enable additional information ──────────────────
Fri Jul 31 18:37:53 2020                                          Press ? for help
```

Figure 14 - Bandwith Monitor of IQ Data Stream

8. The SDR TX output can then be combined into the laboratory plant, and the effect can then be observed in the OFDM channel RxMER data (Figure 15).



2020-08-02 06:27:23

Ref 0 dBm
Att 20 dB

Center Freq 782.4 MHz
Start Freq 680 MHz
Stop Freq 884.8 MHz
Freq Offset 0 Hz
Freq Step 20.48 MHz Auto Manual
Signal Track On Off

Start 680 MHz
Center 782.4 MHz
Stop 884.8 MHz
RBW 300 Hz
VBW 300 Hz
Span 204.8 MHz
SWT 70.31 s

Figure 15 - Impairment Signal on Spectrum Analyzer

In this case we were testing the response of our profile management application (a.k.a. Octave) to verify that proper bitloading values were being set for the impacted frequency ranges. For an example OFDM profile, see Figure 16.

```
    DS      DS  Low-High Freq          Bits/Symbol/  MaxRate
  S/C/CH   Prof Edge (MHz.KHz)    Mod     Subcarrier   Mb/sec
     9/0/48  0    -default-      256qam     8.00     584
    9/0/48  1    -default-      4096qam    11.73     856
     9/0/48  1   702.000-708.000  2048qam      -      -
     9/0/48  1   735.000-737.000  2048qam      -      -
     9/0/48  1   738.000-744.000  2048qam      -      -
     9/0/48  1   745.000-756.000  2048qam      -      -
    9/0/48  2   -default-       4096qam    10.75     785
     9/0/48  2   795.000-801.000  1024qam      -      -
     9/0/48  2   802.000-813.000  512qam       -      -
     9/0/48  2   816.000-828.000  512qam       -      -
     9/0/48  2   831.000-832.000  2048qam      -      -
                           .
                           .
                           .
```

Figure 16 - OFDM Profile for SDR Generated Impairment Signal

9. Equipment Used

Software-Defined Radio Hardware:

1. Ettus Research USRP N320
2. Ettus Research USRP B210 Mini
3. Great Scott Gadgets HackRF One
4. 75 MHz to 1 GHz telescoping antenna
5. SMA Make Right Angle antenna (850/900/1800/1900/2100 MHz, 2 dBi gain, 50 ohm)
6. 10 Gb direct-attach copper cables

Software-Define Radio Software:

1. GNU Radio v3.8.1.0
2. Ettus Research UHD drivers UHD_3.15.0.HEAD-0-gaea0e2de
3. GNU C++ v9.3.0
4. Python 3.8.2
5. Various libraries including libosmodsr-dev, libhackrf-dev, gr-osmosdr, gr-fosphor,

SDR Controller Host Computer:

- CPU - AMD Ryzen Threadripper 3970X
- Motherboard - Asus ROG Zenith II Extreme Alpha
- RAM – 32 GB Corsair DDR4 3600
- Storage – Samsung 970 EVO M.2 1 TB (x2)
- GPU - EVGA GeForce RTX 2080 Ti Xc Hybrid 11 Gb DDR6
- Network – Asus XG-C100C 10G Network Adapter PCI-E x4
- Power – Thermaltake 1200 W 80+ Platinum

OS Software:

1. Ubuntu 20.04 LTS
2. Windows 10

Spectrum Analyzer:

- Siglent SSA 3021X Plus (9 kHz to 2.1 GHz)

10. Conclusion

In this paper, we have explored what software-defined radio is and what it offers to engineers, students, and designers. We presented a general architecture as well as some needed background information on sampling theory and the discrete Fourier transform. Finally, the hardware, software, and drivers used to implement real data flows to sample and/or generate signals were detailed, along with two laboratory use cases.

With the abundance of open-source community supported software and low cost hardware, software-defined radio is proving be a valuable laboratory and field testing tool. By abstracting the functionality from a specific piece of physical hardware, we can transmit and receive various modulation methods, perform signal processing and filtering, and customize routines though Python extensions all within a common software framework. These advantages, combined with the ability to embed Python code for customized signal processing routines, positions software-defined radio as a valuable laboratory and field testing tool.

Abbreviations

ADC	analog-to-digital converter
AWS	advanced wireless service
bps	bits per second
CLI	command line interface
CMTS	cable modem termination system
CPU	central processing unit
DAC	digital-to-analog converter
dB	decibel
dBi	decibel isotropic
DC	direct current
DFT	discrete Fourier transform
DSP	digital signal processing
DTFT	discrete time Fourier transform
FDD	frequency division duplex (or duplexing)
FFT	fast Fourier transform
FPGA	field programmable gate array
GB	gigabyte
GbE	gigabit Ethernet
Gbps	gigabits per second
GHz	gigahertz
GPSDO	Global Positioning System disciplined oscillator
GRC	GNU Radio Companion
GSM	global system for mobile communications (originally *groupe spécial mobile*)
GUI	graphical user interface
Hz	hertz
IEEE	Institute of Electrical and Electronics Engineers
IF	intermediate frequency
IFFT	inverse fast Fourier transform
I/O	input/output
IQ	in-phase/quadrature
ISBE	International Society of Broadband Experts
kHz	kilohertz
LO	local oscillator
LTE	long term evolution
MHz	megahertz
MS/s	megasamples per second
OFDM	orthogonal frequency division multiplexing
OS	operating system
PCS	personal communications service
PMA	profile management application
PPS	pulse per second

QAM	quadrature amplitude modulation
QSFP+	quad small form-factor pluggable plus
RAM	random access memory
RF	radio frequency
RU	rack unit
RX	receive (or receiver)
RxMER	receive modulation error ratio
SCTE	Society of Cable Telecommunications Engineers
SDR	software-defined radio
SFP+	enhanced small form-factor pluggable
SMA	subminiature version A [connector]
SoC	system on a chip
TFTP	Trivial File Transport Protocol
TX	transmit (or transmitter)
UE	user equipment
UHD	USRP hardware driver
USB	universal serial bus
USRP	universal software radio peripheral

Bibliography & References

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4. Cable Modem Operations Support System Interface Specification CM-SP-CM-OSSIv3.1-I17-200610