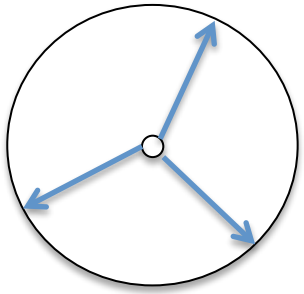
EE194-EE290C

28 nm SoC for IoT

Acknowledgement: Wayne Stark EE455 Lecture Notes, University of Michigan
David Wentzloff EE522 Lecture Notes, University of Michigan

Path Loss



Radiated Power Density [W/m^2]

$$P_{TXdensity} = \frac{P_t}{4\pi R^2}$$

Effective capture area of a receiving antenna

$$A_{RX} = \frac{\lambda^2}{4\pi}$$

Friis transmission formula

$$\mathrm{P_r} = \frac{P_t}{4\pi R^2} A_{RX} = P_t \left(\frac{\lambda}{4\pi R} \right)^2$$

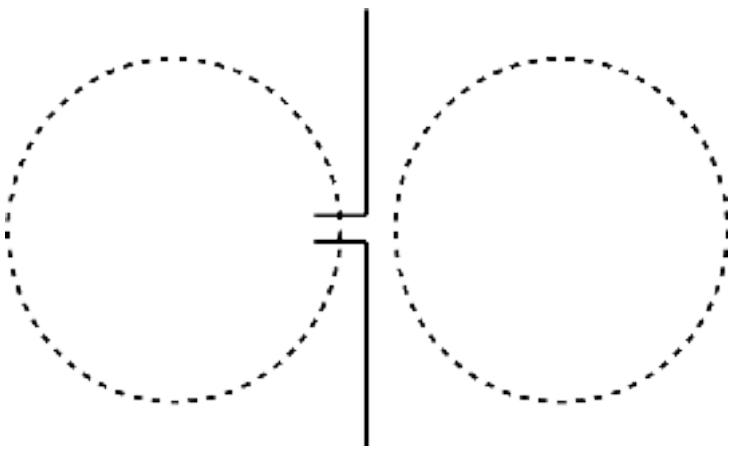
$$\frac{\mathrm{P_r}}{P_t} = \left(\frac{\lambda}{4\pi R} \right)^2$$

Antenna Gain

For the point radiator

$$\frac{\mathrm{P_r}}{P_t} = \left(\frac{\lambda}{4\pi R}\right)^2$$

Antennas have non-spherical radiation patterns



Antenna Gain (G)
Directions where power density higher
Than that of point radiator

Friis transmission formula incorporating radiation pattern

$$\frac{\mathrm{P_r}}{P_t} = G_t G_r \left(\frac{\lambda}{4\pi R}\right)^2$$

Link Budget

- A link budget is used to evaluate system performance

- Accounts for the following
  - Antenna radiation (gain)
  - Path loss
  - Circuit performance (NF)
  - Architecture performance (E_b/N_o)

- Calculates if there is sufficient SNR at the receiver for reliable communication
  - Excess SNR is called “link margin”

Link Budget

Friis Equation:

$$\frac{\mathrm{P_r}}{P_t} = G_t G_r \left(\frac{\lambda}{4\pi R} \right)^2$$

$$\mathrm{P_r} = P_t + G_t + G_r + 20\log_{10}\left(\frac{\lambda}{4\pi R} \right)$$

Gain has units of dB and power has units of dBm or dBW.

Link Budget

P_t (Transmit power)	-5 dBm (316.2µW)
G_t (Transmit antenna Gain)	0 dB
G_r (Receive antenna Gain)	0 dB
λ (2.45 GHz)	0.12491 m
P_r (Received power)	-75 dBm (31.6pW)
Link Margin	70 dB
Distance (d)	~17 m

$$NF < R_{ss} - SNR_{\min} + 174 - 10\log_{10}(BW)$$

Receiver NF

$$NF < R_{ss} - SNR_{\min} + 174 - 10\log_{10}(BW)$$

For BLE: $P_e = 0.1\% = 10^{-3}$

$$SNR = \frac{Signal}{Noise} = \frac{E_b \times R}{N_o \times B} = \frac{E_b}{N_o}$$

E_b/N_o depends on the type of modulation and demodulator.

Matlab Signal & Noise Modeling

According to the sampling theorem we can represent **x(t)** by samples at rate $\mathbf{2f_{max}}$.

In addition, assume that **x(t)** is causal so that x(t) = 0 for t < 0.

The Fourier transform of x(t) is given by

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt$$

$$= \int_{0}^{\infty} x(t) e^{-j2\pi ft} \, dt$$

Notice that when **x(t)** is real then **X(−f) = X*(f)**.

Matlab Signal & Noise Modeling

The inverse Fourier transform is

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft}\, df$$

$$= \int_{-f_{\max}}^{f_{\max}} X(f) e^{j2\pi ft}\, df$$

Now we can approximate these integrals with finite sums.

Matlab Signal & Noise Modeling

- Since the highest frequency is $\mathbf{f_{max}}$ we sample at a rate $\mathbf{2f_{max}}$.
- Thus the samples are separated in time by $\mathbf{\Delta t = 1/(2f_{max})}$.
- Suppose we have **N** samples in the time domain.
- We can reconstruct these **N** samples in the time domain with just **N** samples in the frequency domain.
- The spacing in the frequency domain is $\mathbf{\Delta f = 2f_{max}/N}$.
- Since $\mathbf{\Delta t = 1/(2f_{max})}$ and $\mathbf{\Delta f = 2f_{max}/N}$

$$\Delta f \Delta t = \frac{1}{N}$$

- That is, we are dividing the interval of the frequency response from $\mathbf{-f_{max}}$ to $\mathbf{f_{max}}$ into **N** discrete frequencies and we are dividing time from **0** to $\mathbf{(N-1)\,\Delta t}$ into **N** discrete time samples.

Matlab Signal & Noise Modeling

Then for **$|f|<f_{max}$**

$$X(f) = \int_0^{\infty} x(t) e^{-j2\pi ft}\, dt$$

$$\approx \sum_{l=0}^{N-1} x(l\Delta t) e^{-j2\pi fl\Delta t} \Delta t$$

$$X_m = X(m\Delta f) = \Delta t \sum_{l=0}^{N-1} x(l\Delta t) e^{-j2\pi m\Delta f l\Delta t}$$

$$= \Delta t \sum_{l=0}^{N-1} x(l\Delta t) e^{-j2\pi ml/N}$$

Matlab Signal & Noise Modeling

- Thus the above expression for $\mathbf{X_m}$ is valid for $\mathbf{0 \le |m| \le N/2}$.
- Notice that $\mathbf{X_{-m} = X_{N-m}}$.
- In MATLAB the operation FFT takes an input vector (indexed from 1 to N) and produces an output vector (indexed from 1 to N).
- The input-output relation is

$$Xk = \sum_{n=1}^{N} x_n e^{-j2\pi(k-1)(n-1)/N} \qquad k = 1, \ldots, N$$

The input should correspond to the samples starting at time **0** up to time **(N – 1)Δt**. Notice that in order to get the correct magnitude of the frequency response we must multiply what MATLAB produces by **Δt**.

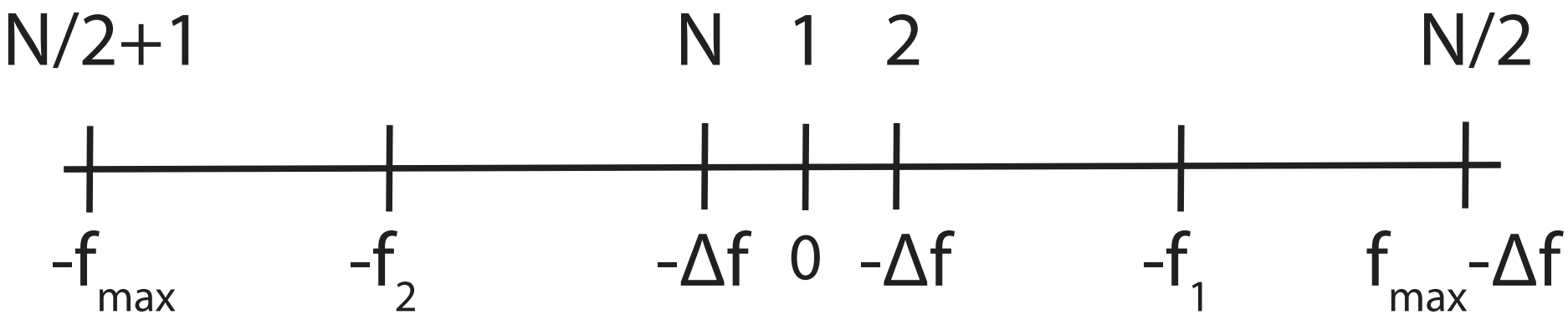
Matlab Signal & Noise Modeling

- The output samples from 1 to **N/2** (when normalized appropriately) correspond to the frequency response **$X(0), X(\Delta f), \ldots, X((N/2)\Delta f) = X(f_{max} - \Delta f)$**.
- The samples from **N/2 + 2** to **N** correspond to the frequency response from **$X(-f_{max} + \Delta f), \ldots X(-\Delta f)$**.
- The order can be reversed into the logical order with the MATLAB command fftshift.

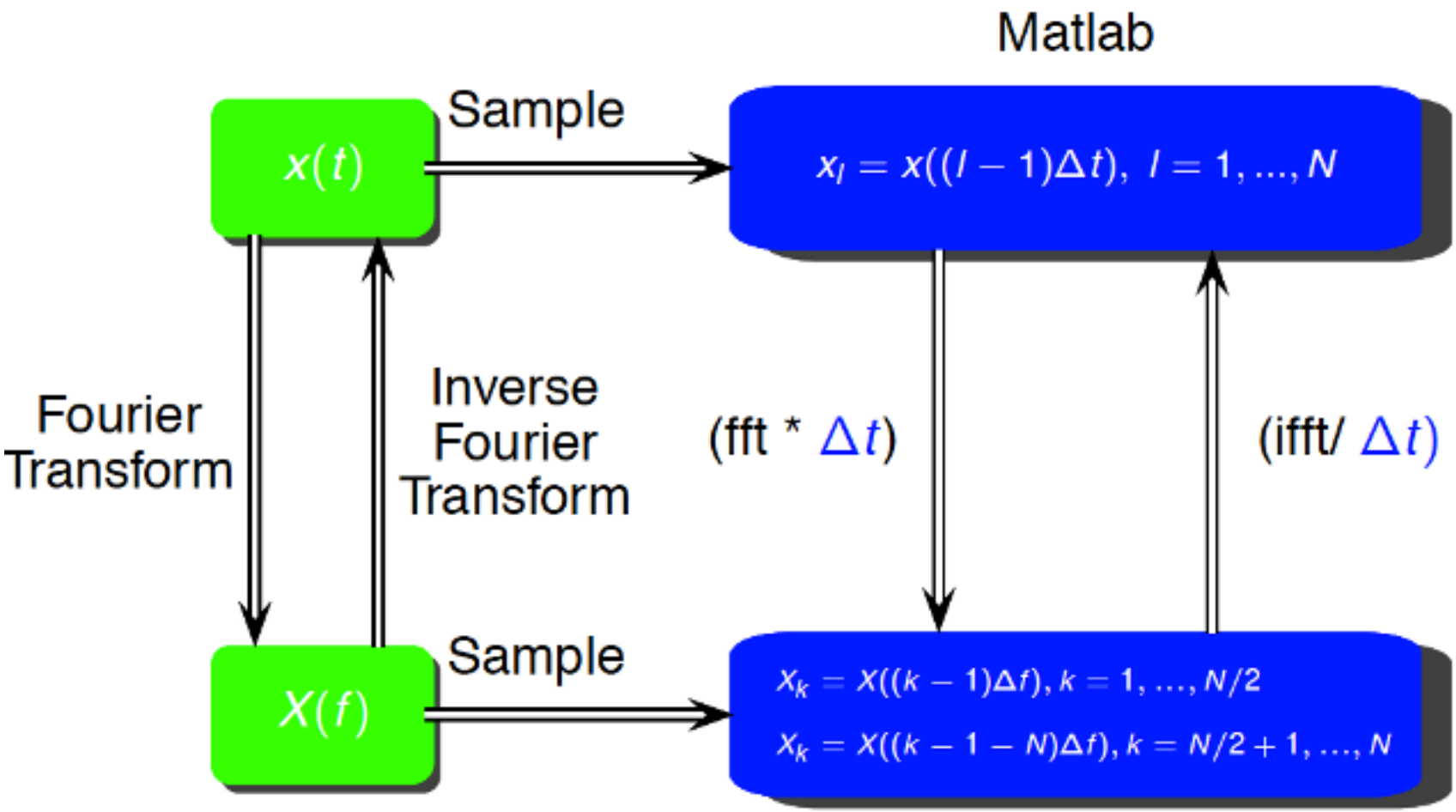
Matlab Signal & Noise Modeling

Matlab Index	Frequency	Matlab Index	Frequency
1	0	N/2+1	$-f_{max}$
2	Δf	N/2+2	$-f_{max}+\Delta f$
3	$2\Delta f$	…	…
…	…	N/2+k	$-f_{max}+(k-1)\ \Delta f$
m	$(m-1)\ \Delta f$	…	…
…	…	N/2+(fmax-f2)/Δf +1	$-f_2$
(f1/Δf)+1	f_1	…	…
…	…	N-1	$-2\Delta f$
N/2	$f_{max}-\Delta f$	N	$-\Delta f$

Matlab Signal & Noise Modeling



Matlab FFT Caveat



Simulating Noise

- In a computer simulation of a communication system we represent a continuous time signal with samples according to the sampling theorem.
- A signal can be represented if the highest frequency content is less than half the sampling rate.
- Thus if $\mathbf{f_s}$ represents the sampling rate the simulation bandwidth is $\mathbf{f_s/2}$.
- If we consider white Gaussian noise the noise has equal power at all frequencies up to the simulation bandwidth.

Simulating Noise

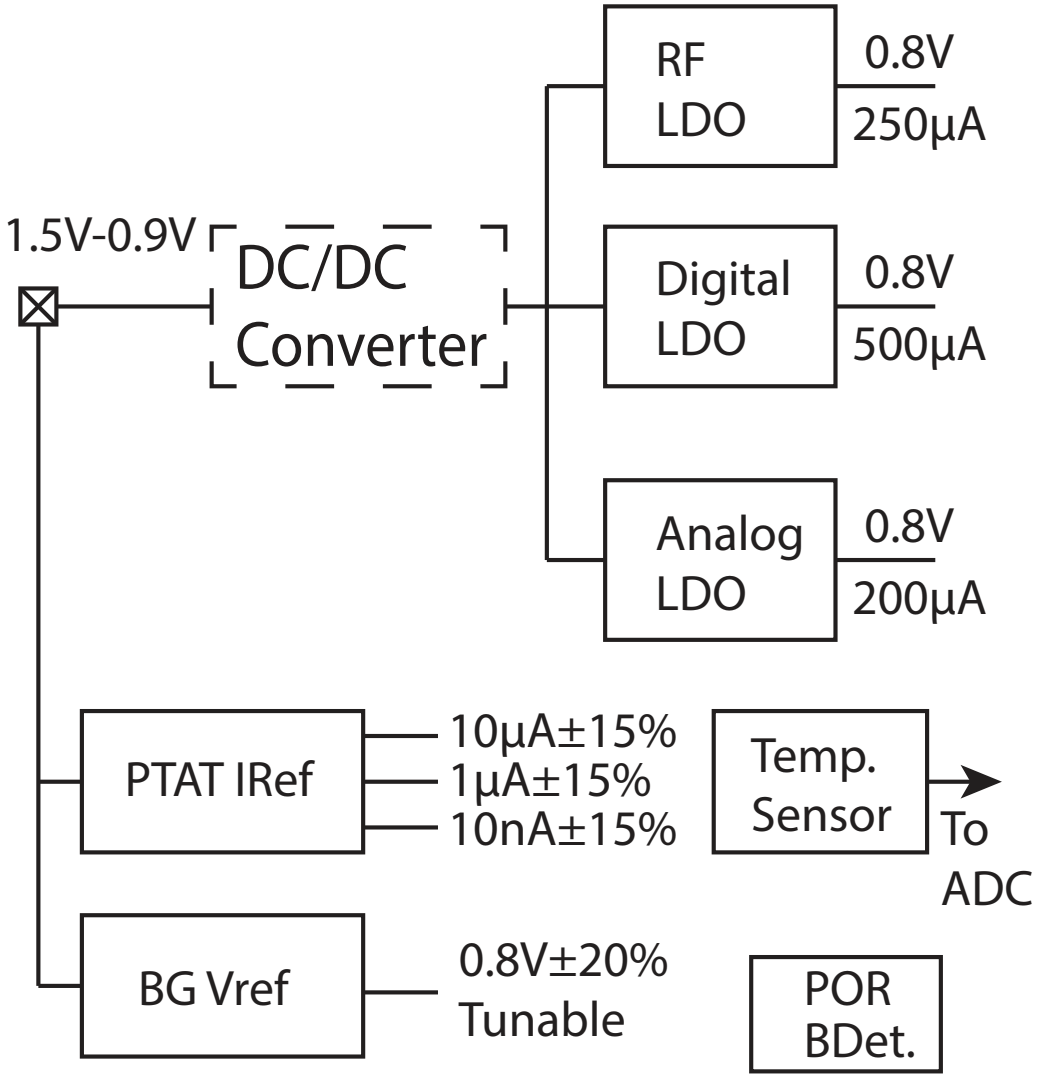
- Thus the power spectral density of the noise in the simulation is

$$S_n(f) = N_0 / 2 \qquad -f_s / 2 < f < f_s / 2$$

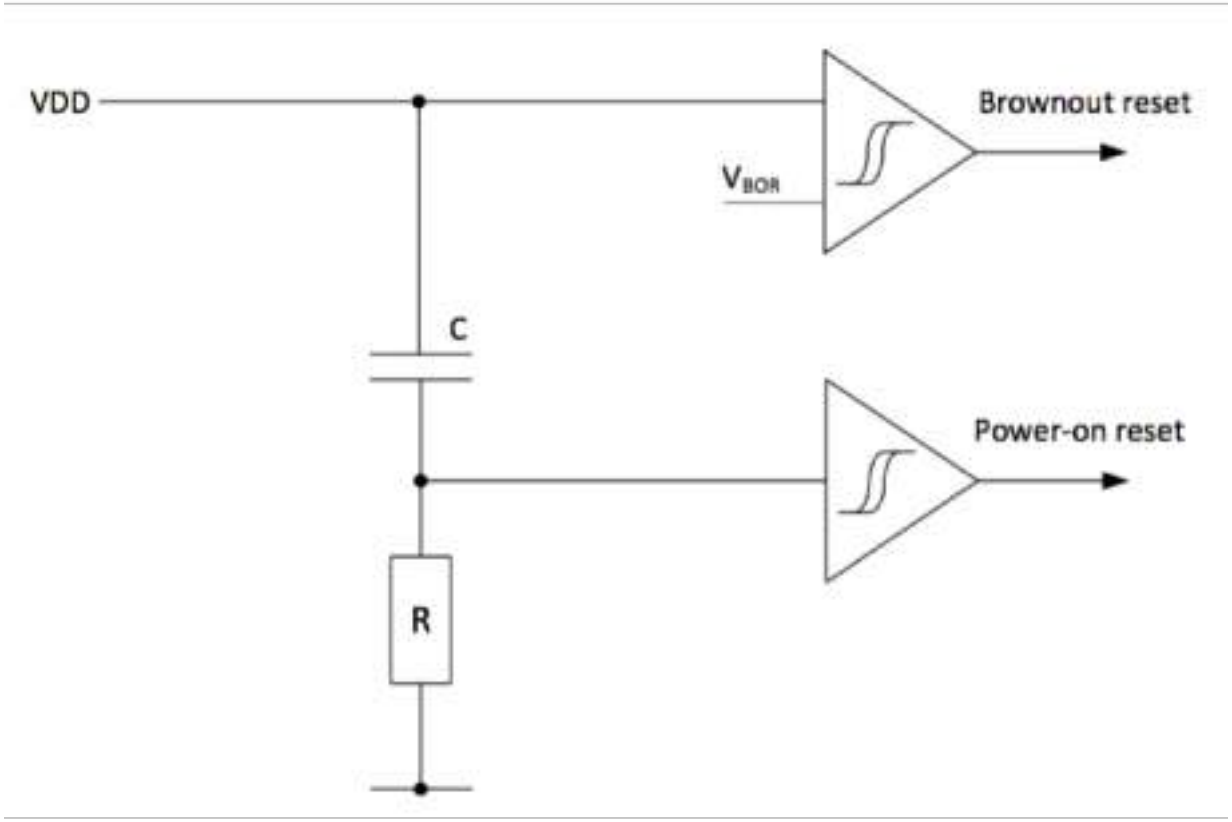
- The noise samples (with this finite bandwidth) have variance

$$\sigma^2 = \int_{-f_s/2}^{f_s/2} S_n(f)\,df = \int_{-f_s/2}^{f_s/2} \frac{N_o}{2}\,df = N_o f_{\max}$$

Power Management



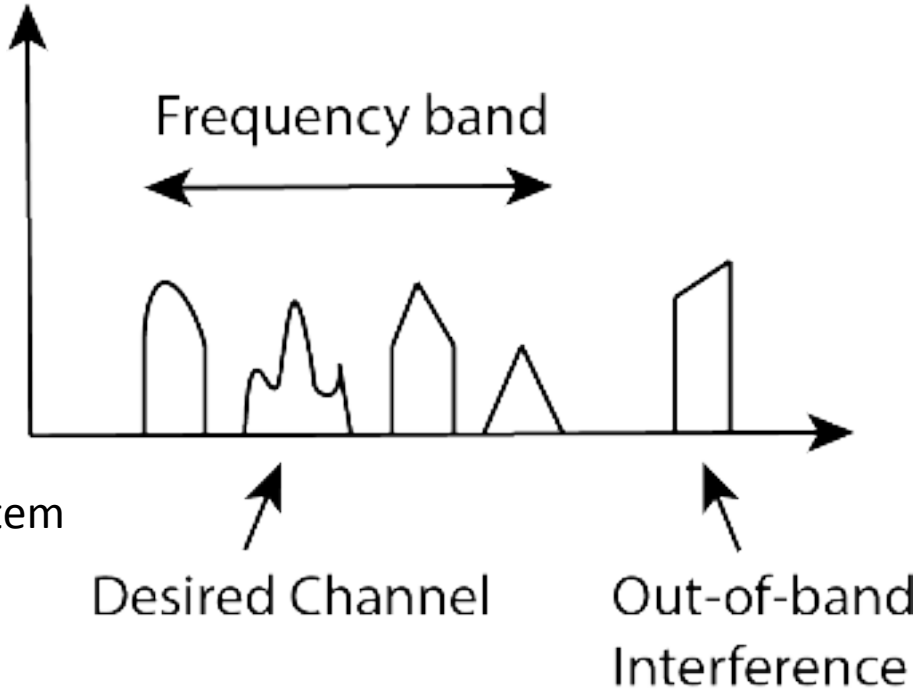
Reset Signals



Nordic: nRF52840 Datasheet.

Terminology

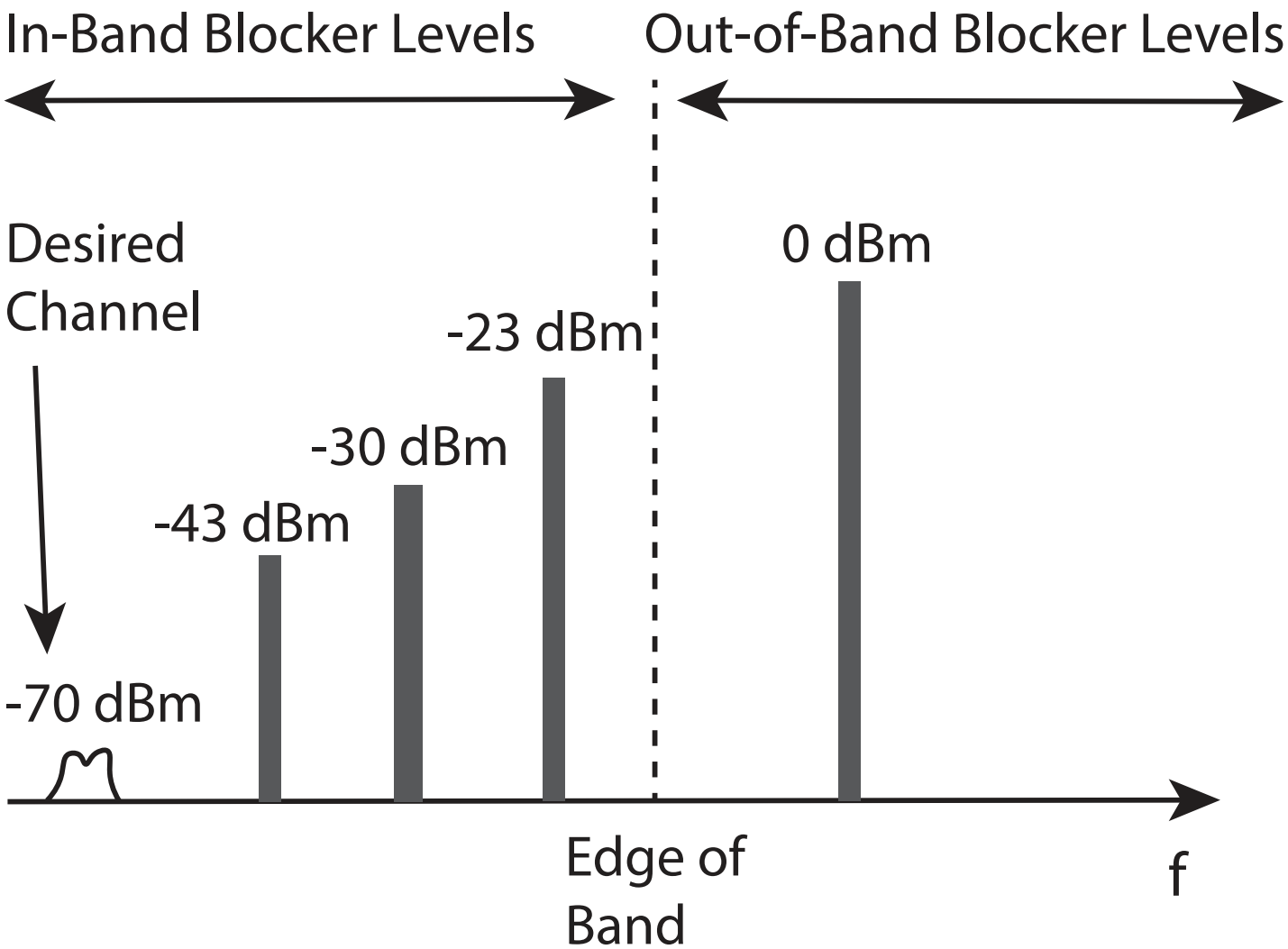
- Band - entire spectrum for a standard
  - GSM 935-960 MHz
  - FM 87.5-108 MHz



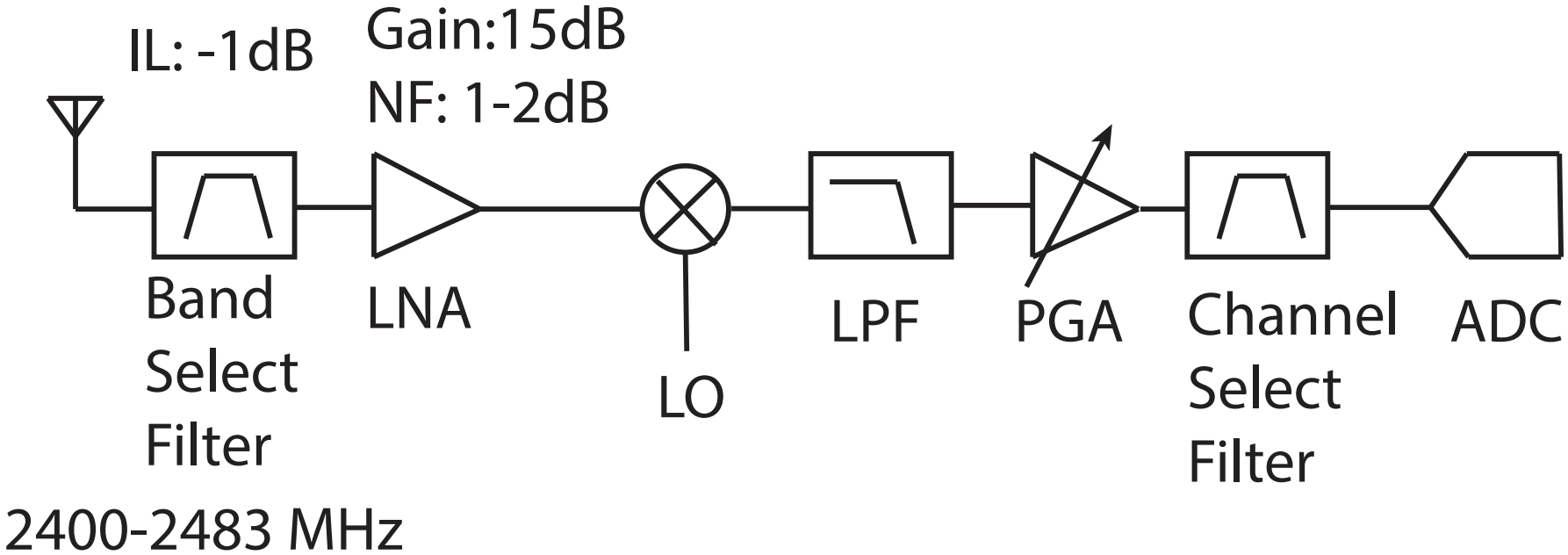
- Channel – bandwidth of 1 user in the system
  - GSM 200 kHz
  - FM 100 kHz

- Band-select filter
- Channel select filter
- Out of band Interference
- In-band Interference
- Adjacent-channel interference
- Alternate-channel interference

Terminology



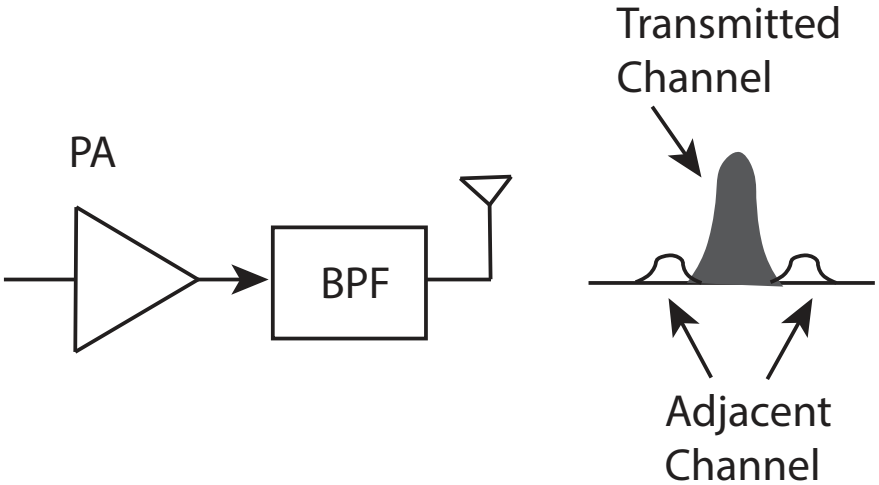
Rx Signal Flow

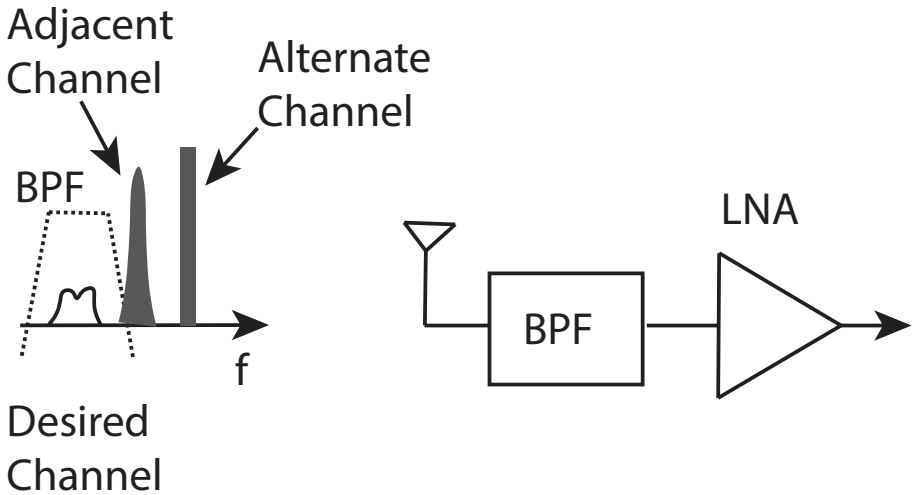


$$F_{total} = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \frac{F_4 - 1}{G_1 G_2 G_3} + \ldots$$

RF Filtering

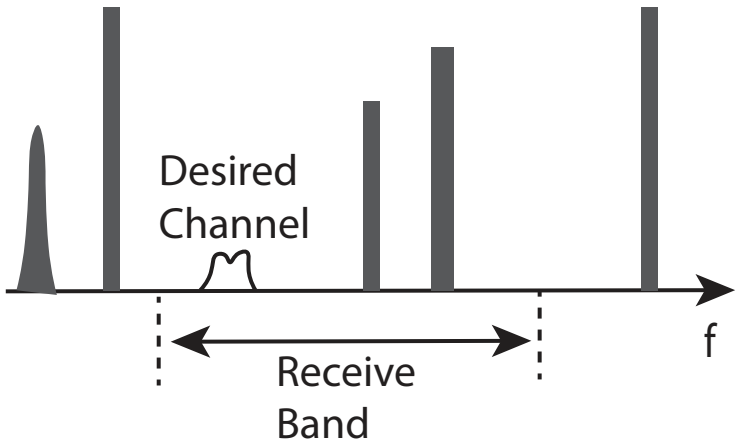
Filtering requirements impact the RF design of the transceiver.





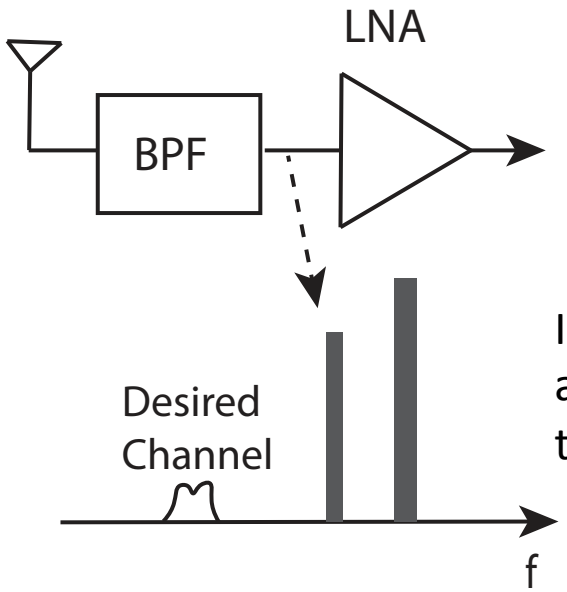
RF Filtering

- Receive chain preceding channel-selection filtering must be sufficiently linear.
- Channel selection filtering is extremely difficult at the RF frequencies.



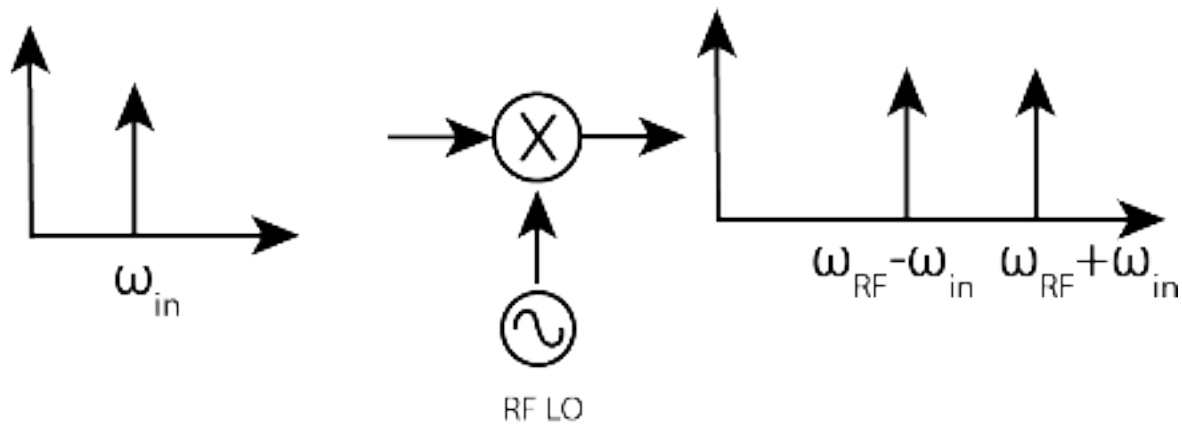
$$Q_{BPF} \approx \frac{f_o}{BW}$$

=2450MHz/1MHz
=2450

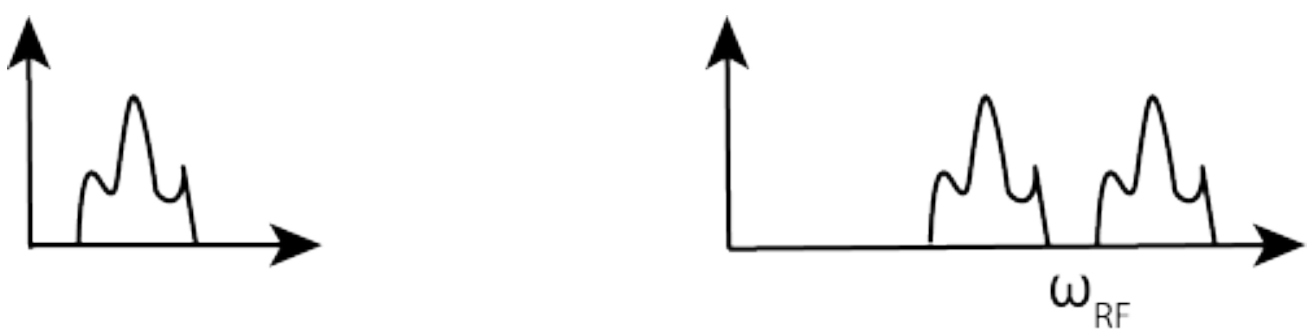


In-Band interferers
are typically removed near
the end of the receive chain.

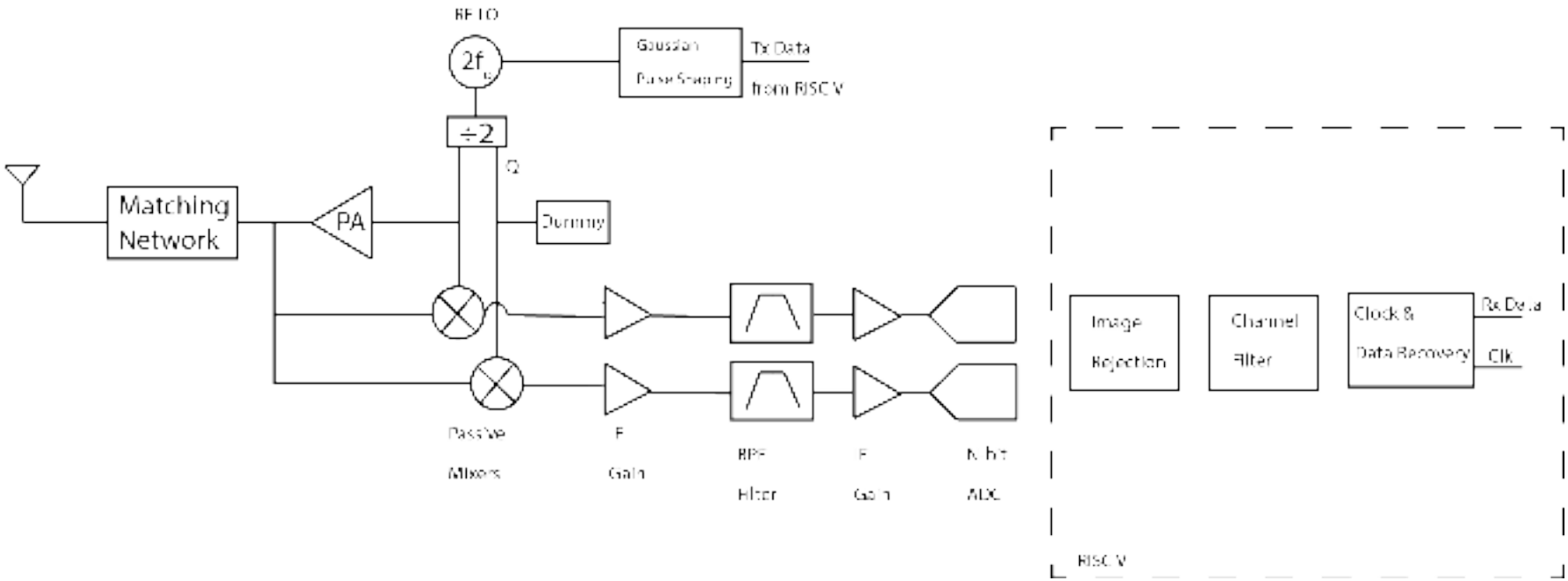
Frequency Translation Mixers



$$\left(A\cos\omega_1 t\right)\left(B\cos\omega_2 t\right)=\frac{AB}{2}\left[\cos\left(\omega_1-w_2\right)t+\cos\left(\omega_1+w_2\right)t\right]$$

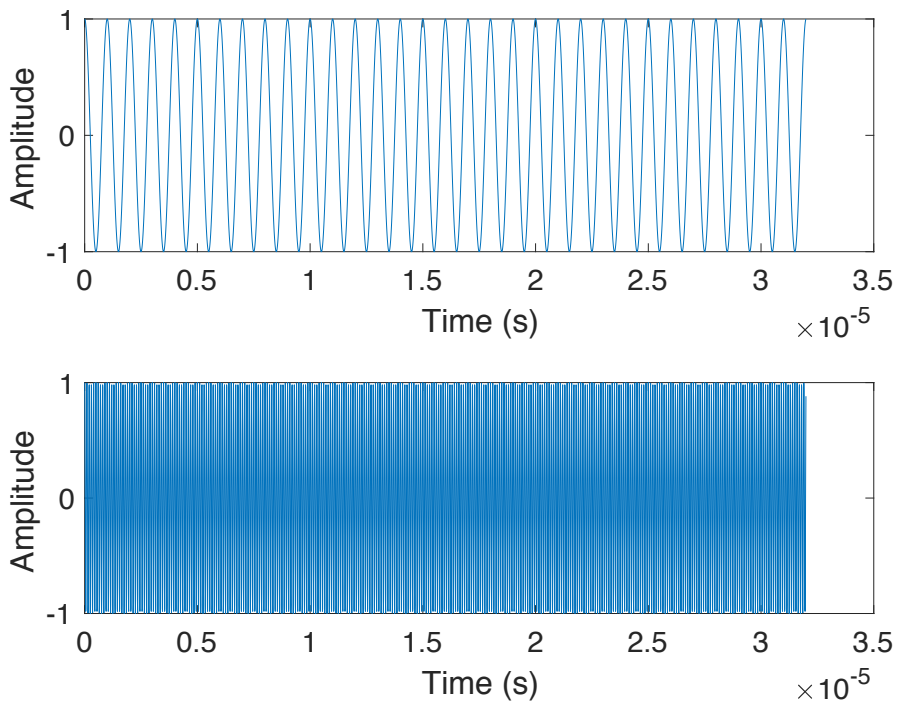


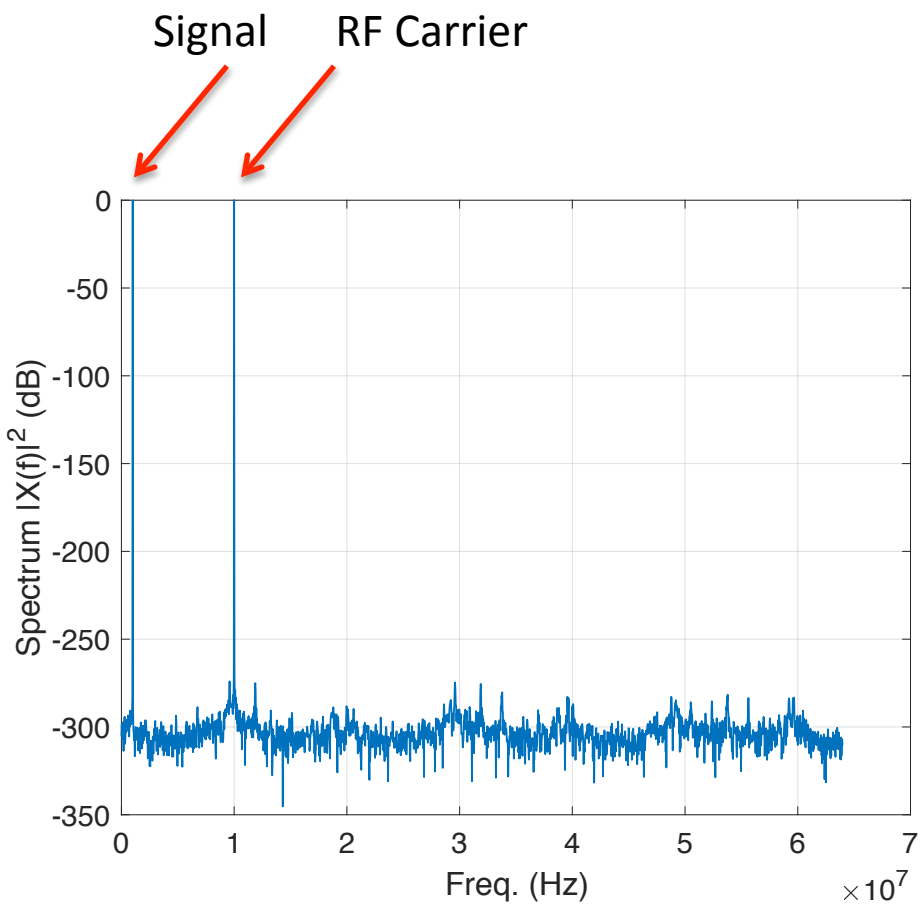
TRx


RF LO
2f
÷2
Q
Gaussian
Pulse Shaping
Tx Data
from RISC V
Matching
Network
PA
Dummy
Passive
Mixers
Gain
BPF
Filter
Image
Rejection
Channel
Filter
Clock &
Data Recovery
Clk
RISC V

Signal Flow

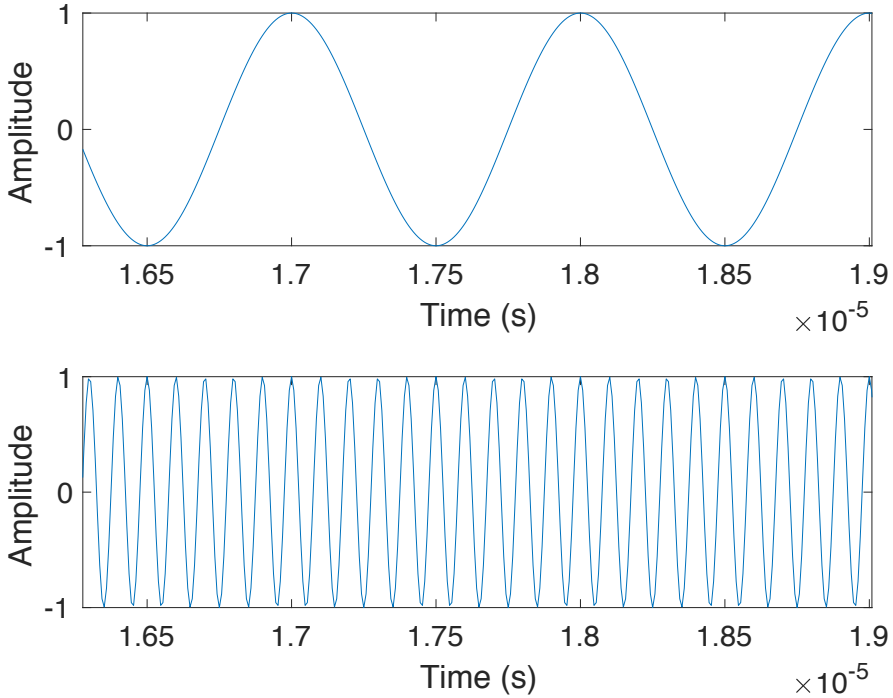
Fs = 128 MHz
N = 2^12
Time = 32 µS

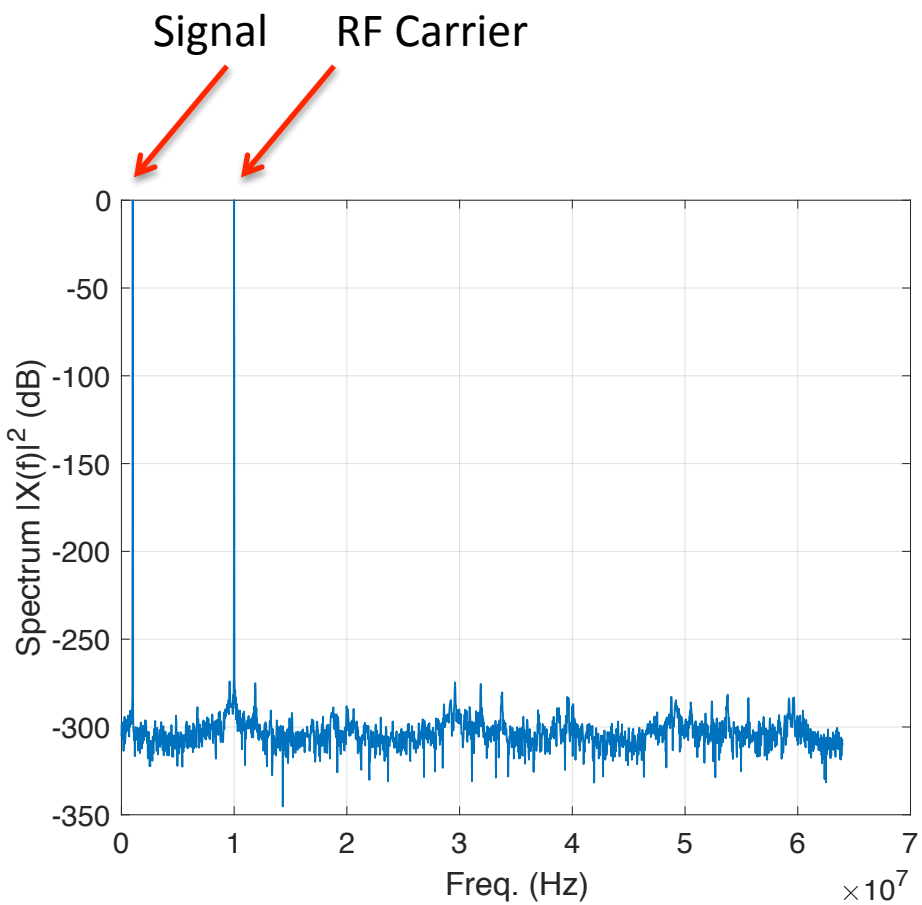




Baseband Signal

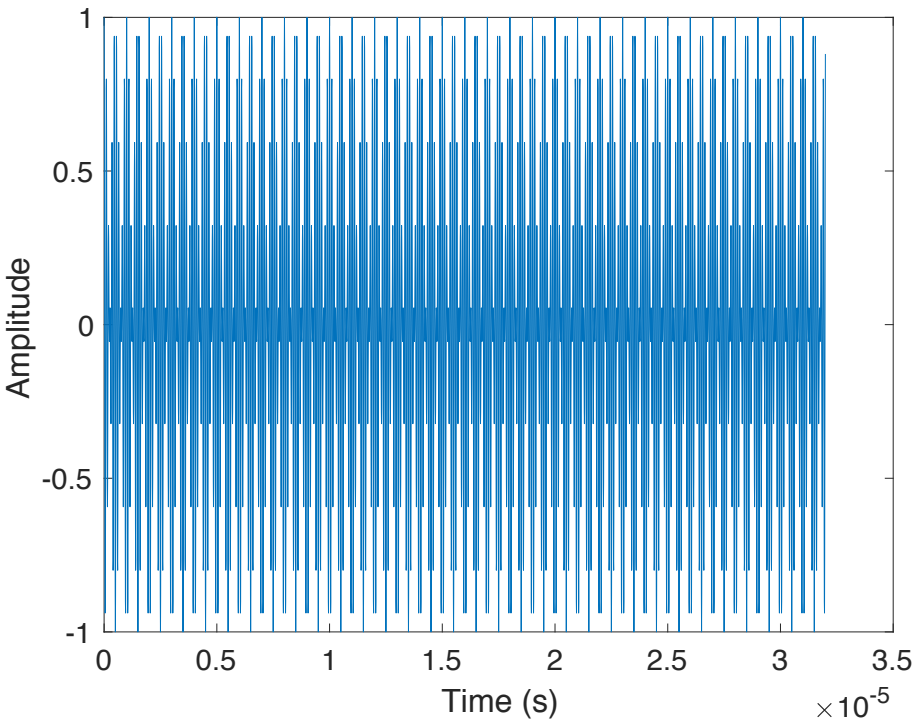
Fs = 128 MHz
N = 2^12
Time = 32 µS

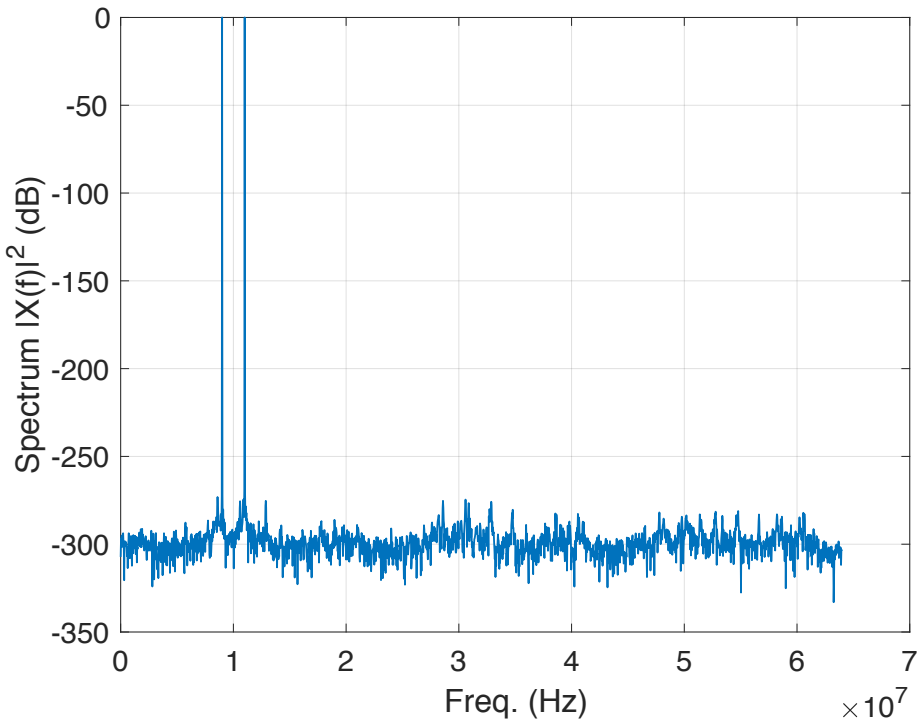




Up Conversion

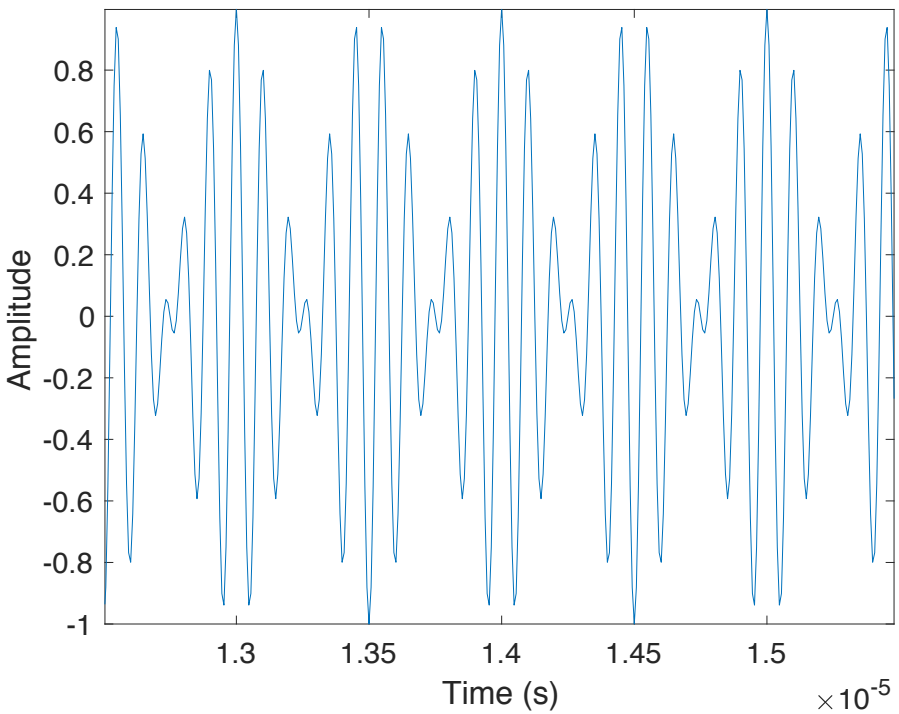
$$x(t) * LO(t)$$

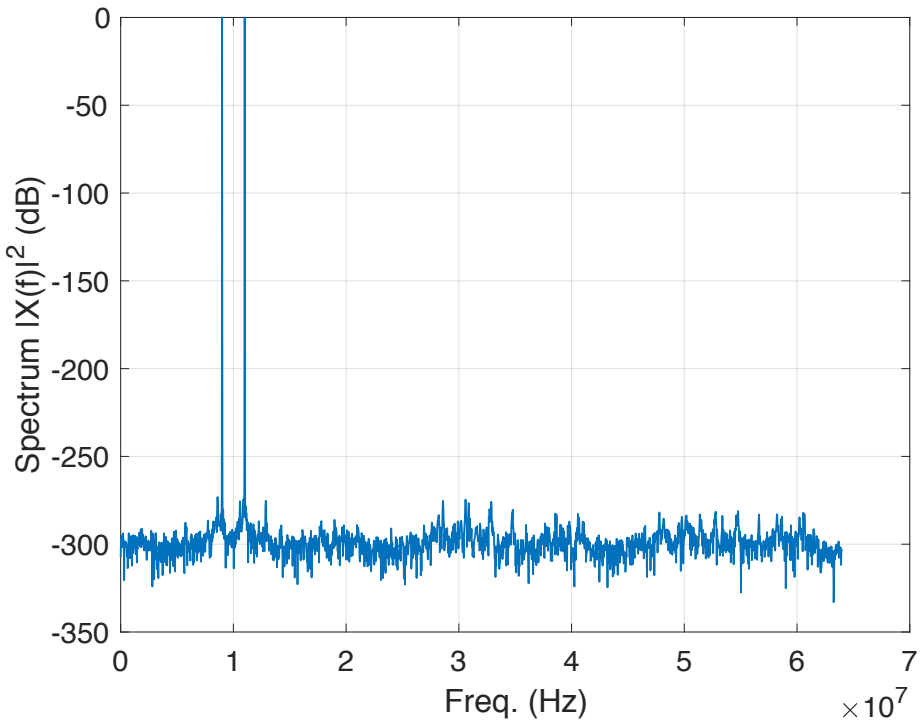




Up Conversion

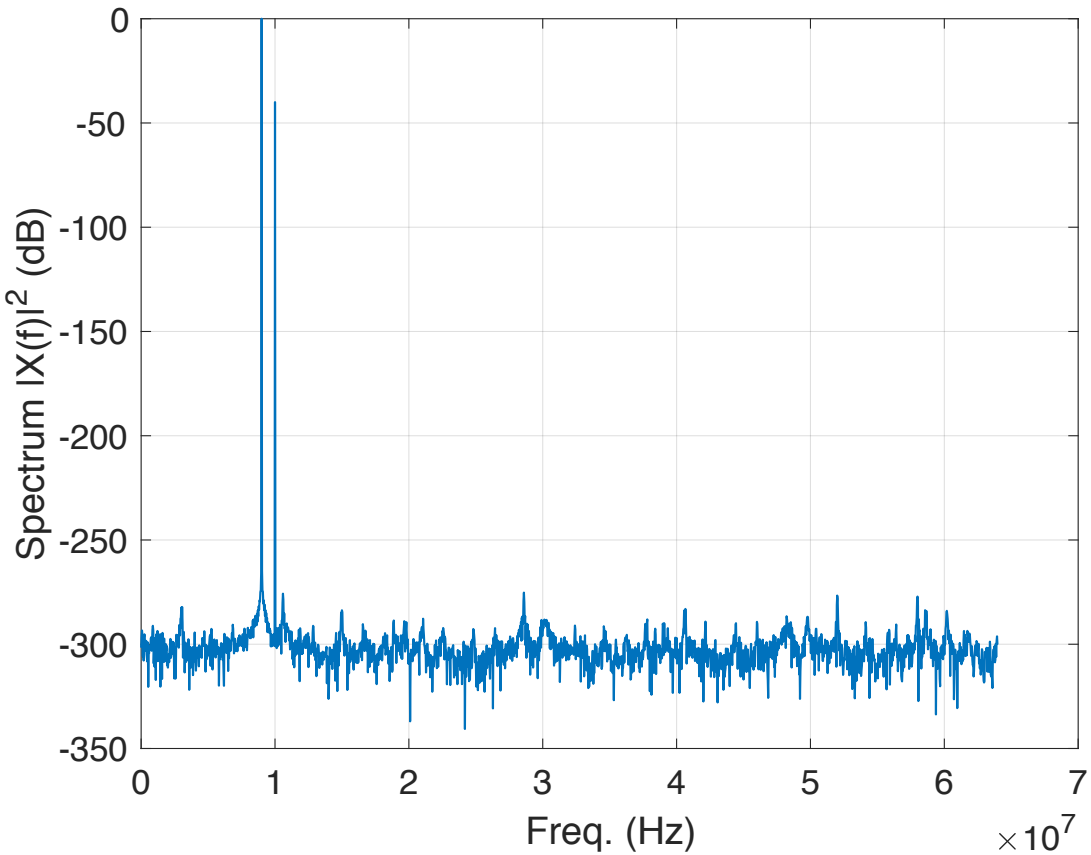
$$x(t) * LO(t)$$





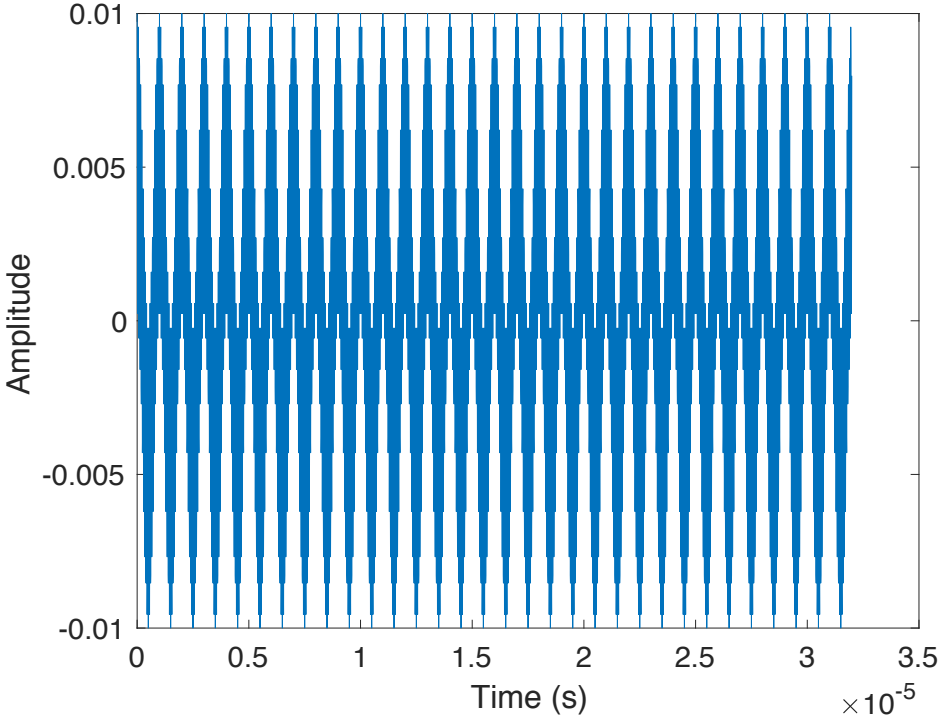
Down Conversion

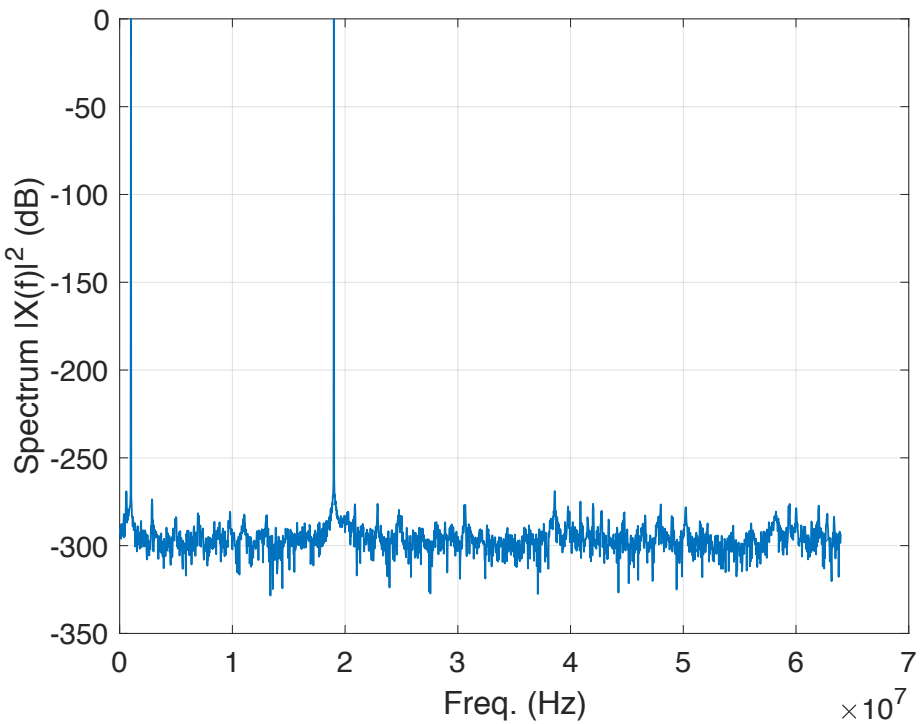
$RF(t) * LO(t)$



Down Conversion

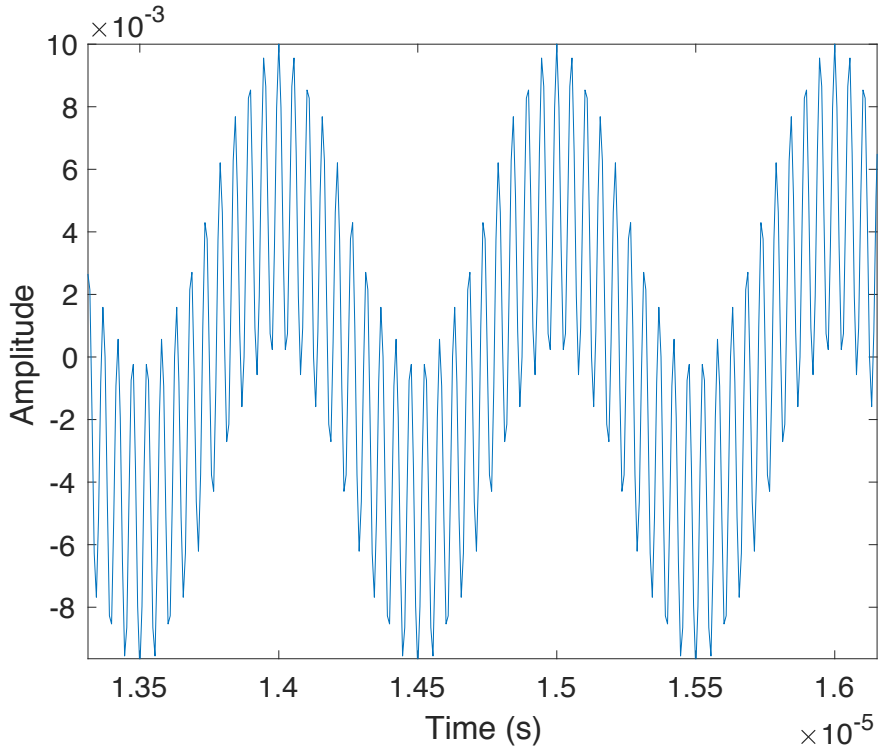
$$RF(t) * LO(t)$$

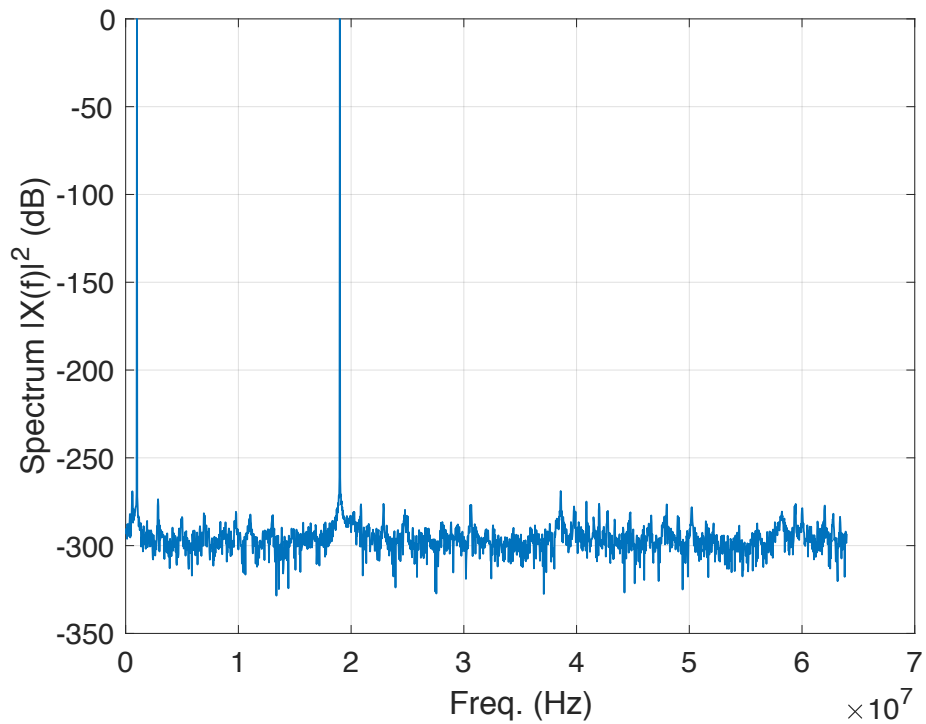




Down Conversion

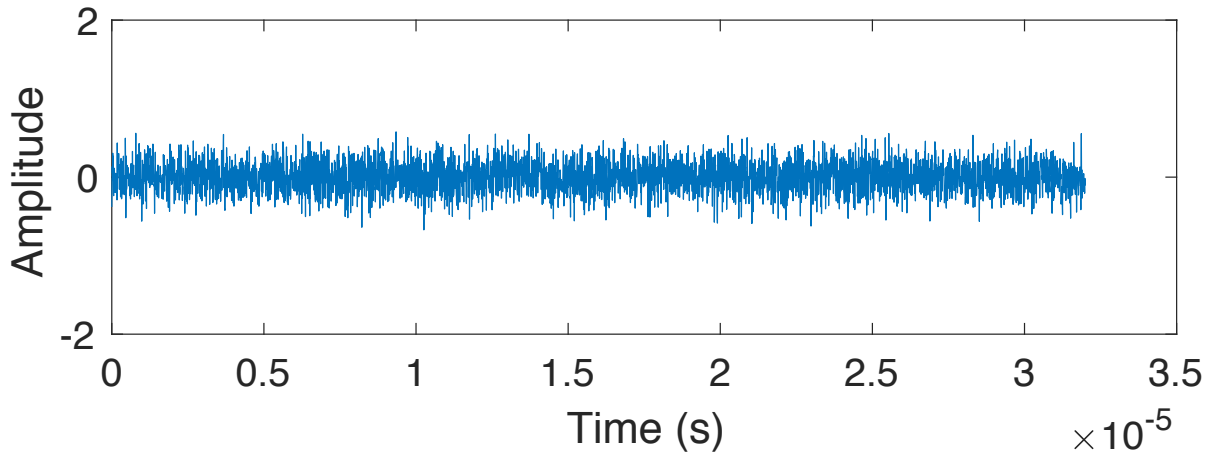
$$RF(t) * LO(t)$$

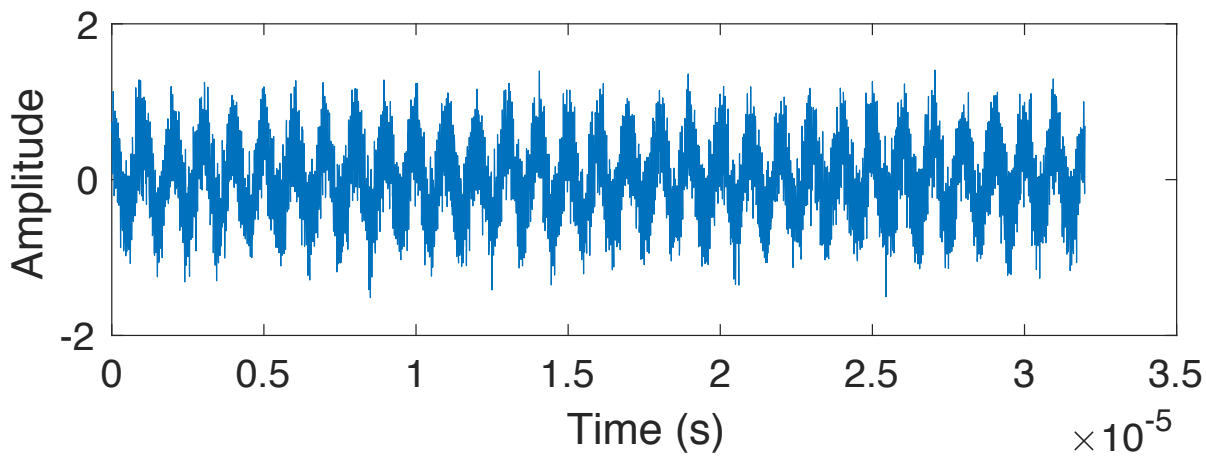




AWGN

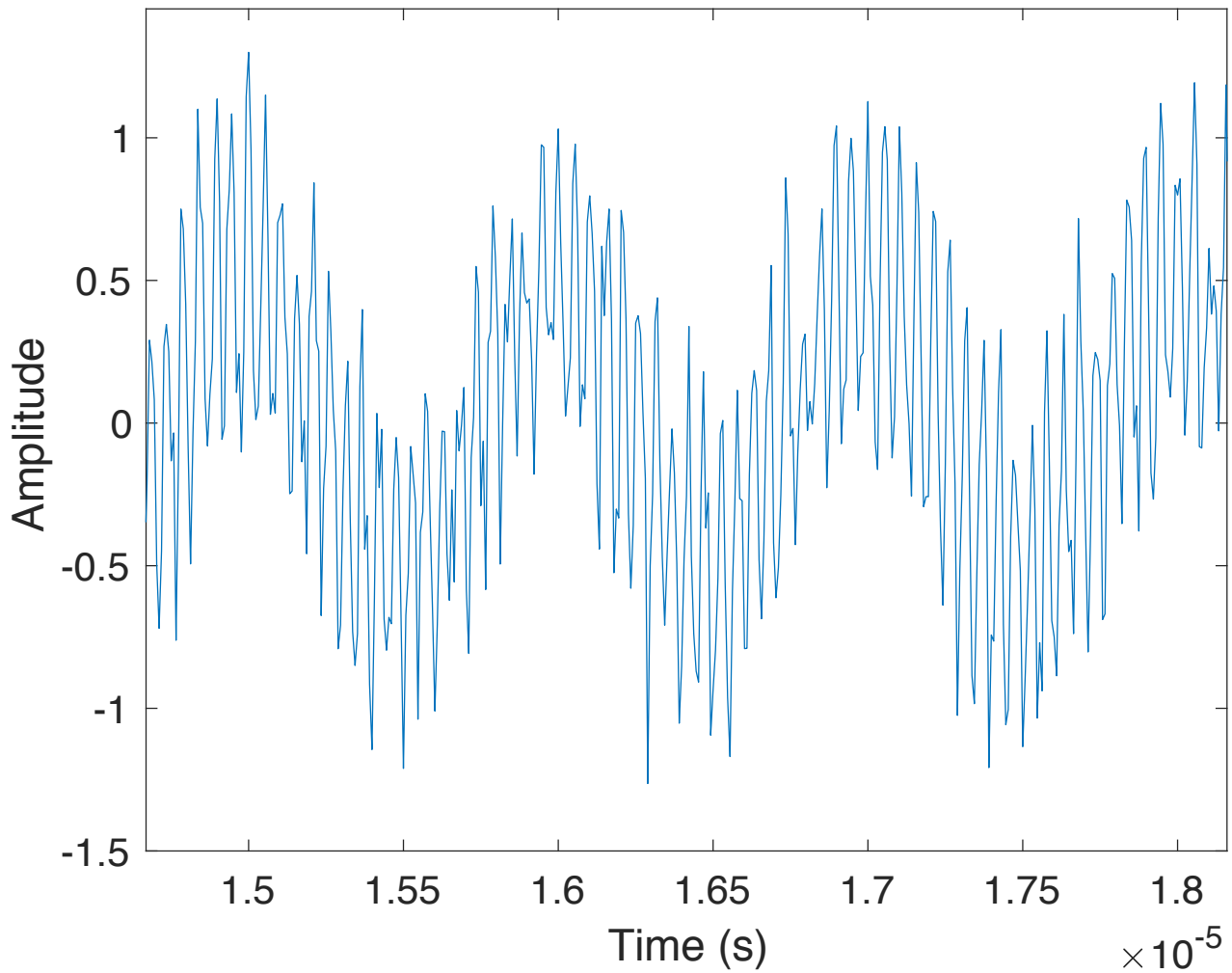
$$RF(t) * LO(t) + n(t)$$





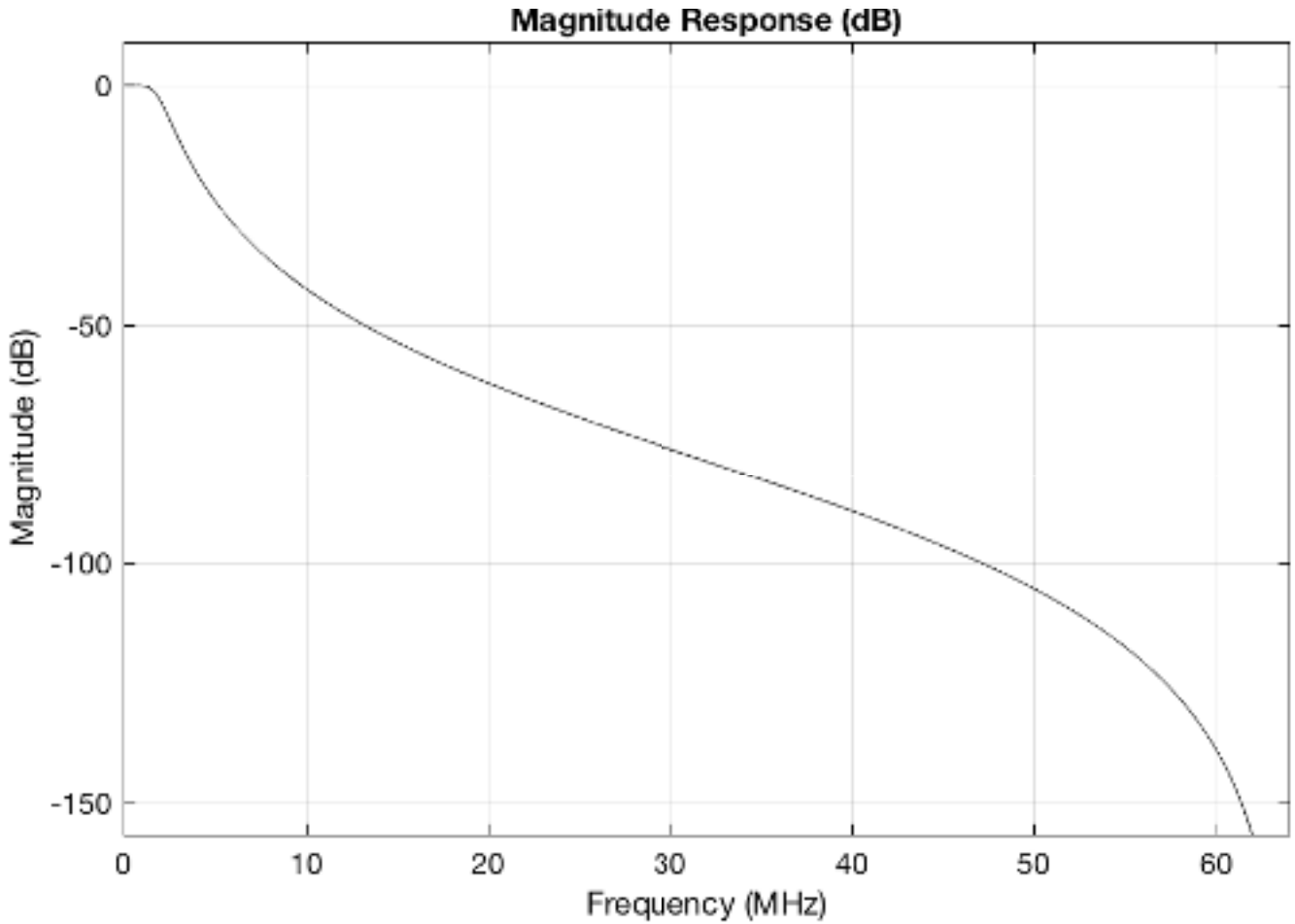
Down Conversion

$$RF(t) * LO(t) + n(t)$$



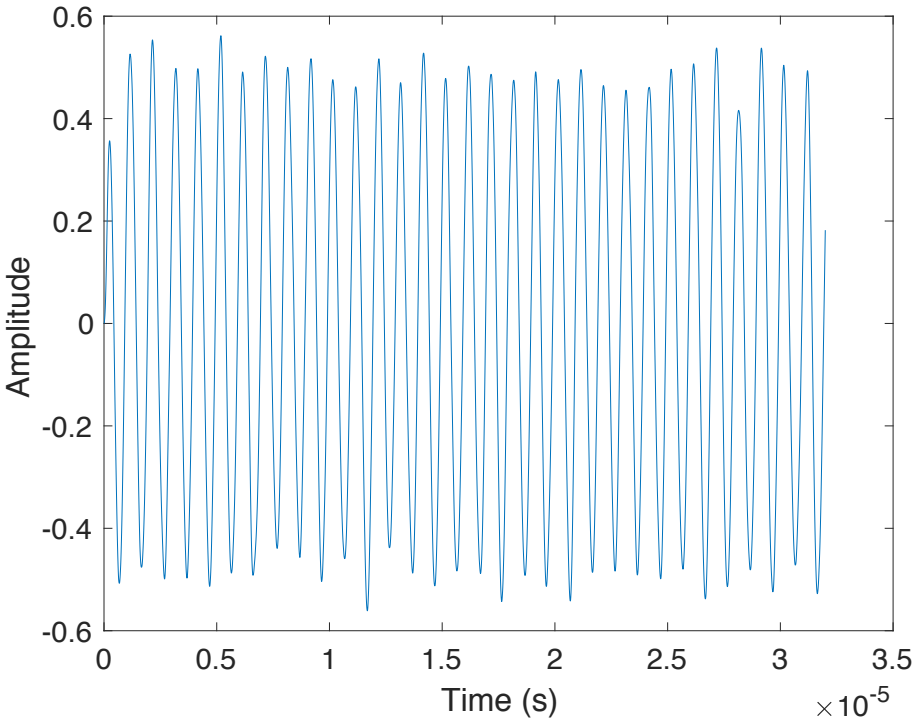
LPF

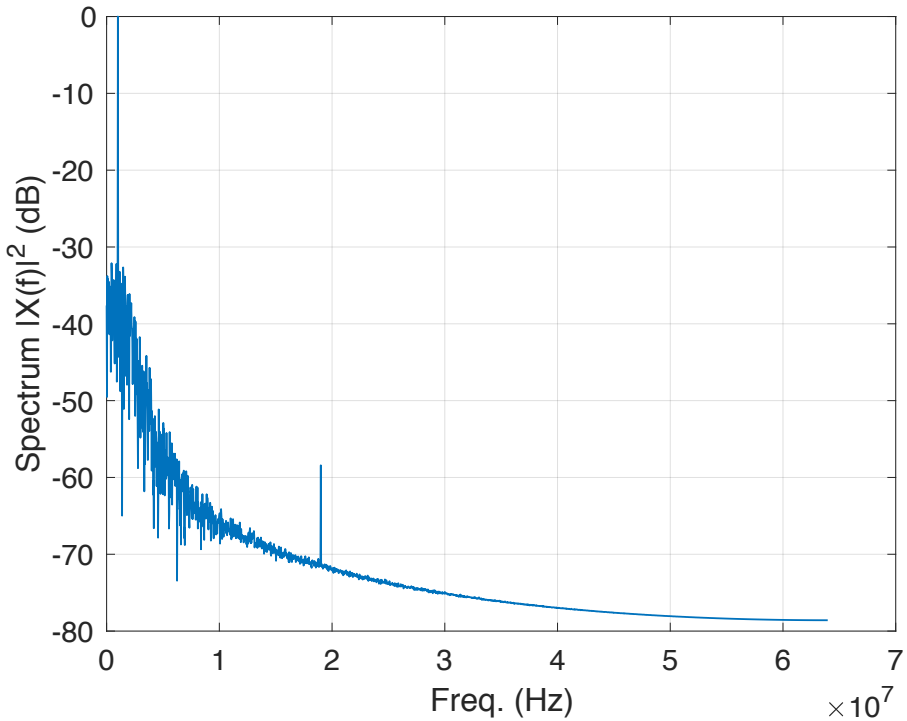
3rd order Butterworth
Fc = 2 MHz



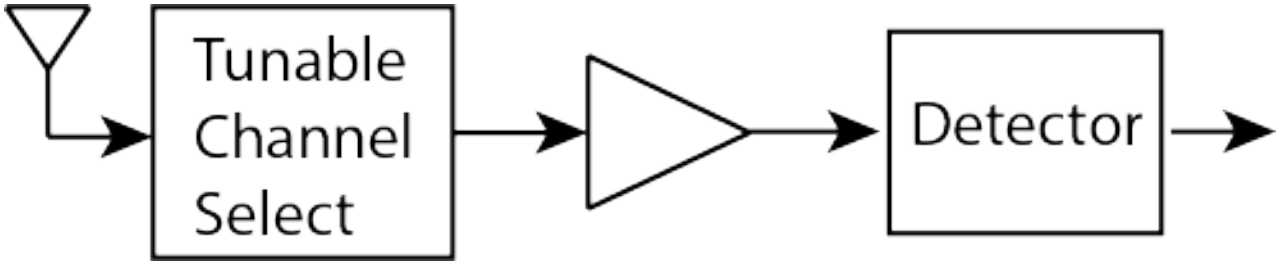
Down Conversion

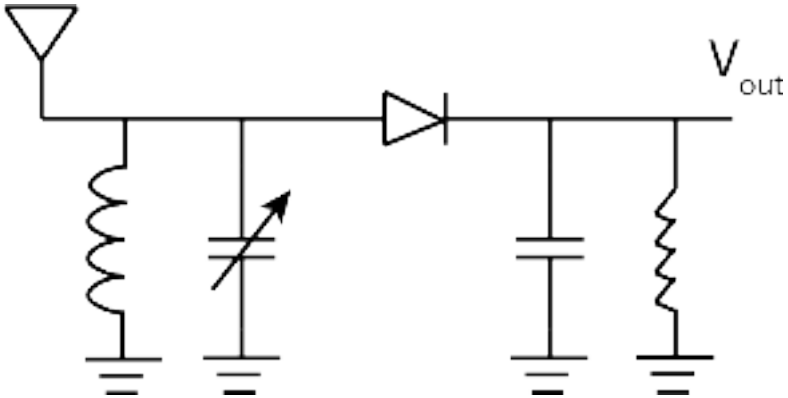
$$RF(t) * LO(t) + n(t)$$





Rx Architecture





- Disadvantages
  - High-Q RF, tunable RF filter required
  - Amplifier should also be tunable, high Q