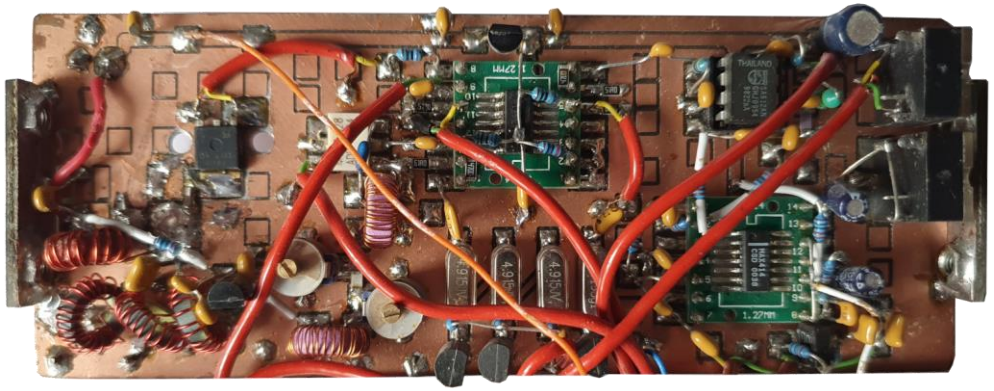
Extra License Course

Session 6 – Chapter 6 Radio Circuits & Systems

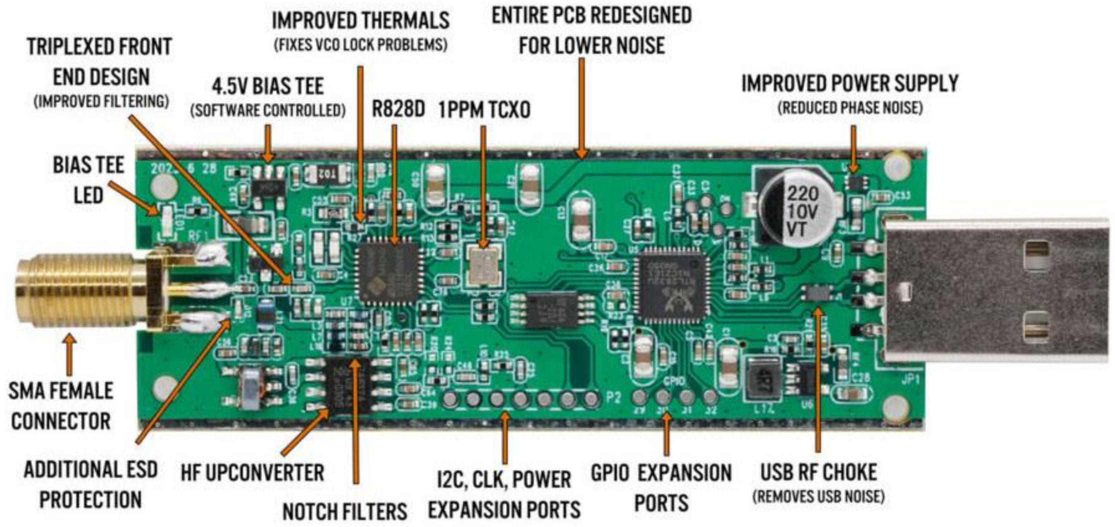
6.3 Digital Signal Processing (DSP) & Software Defined Radio (SDR)

6.3 DSP & SDR

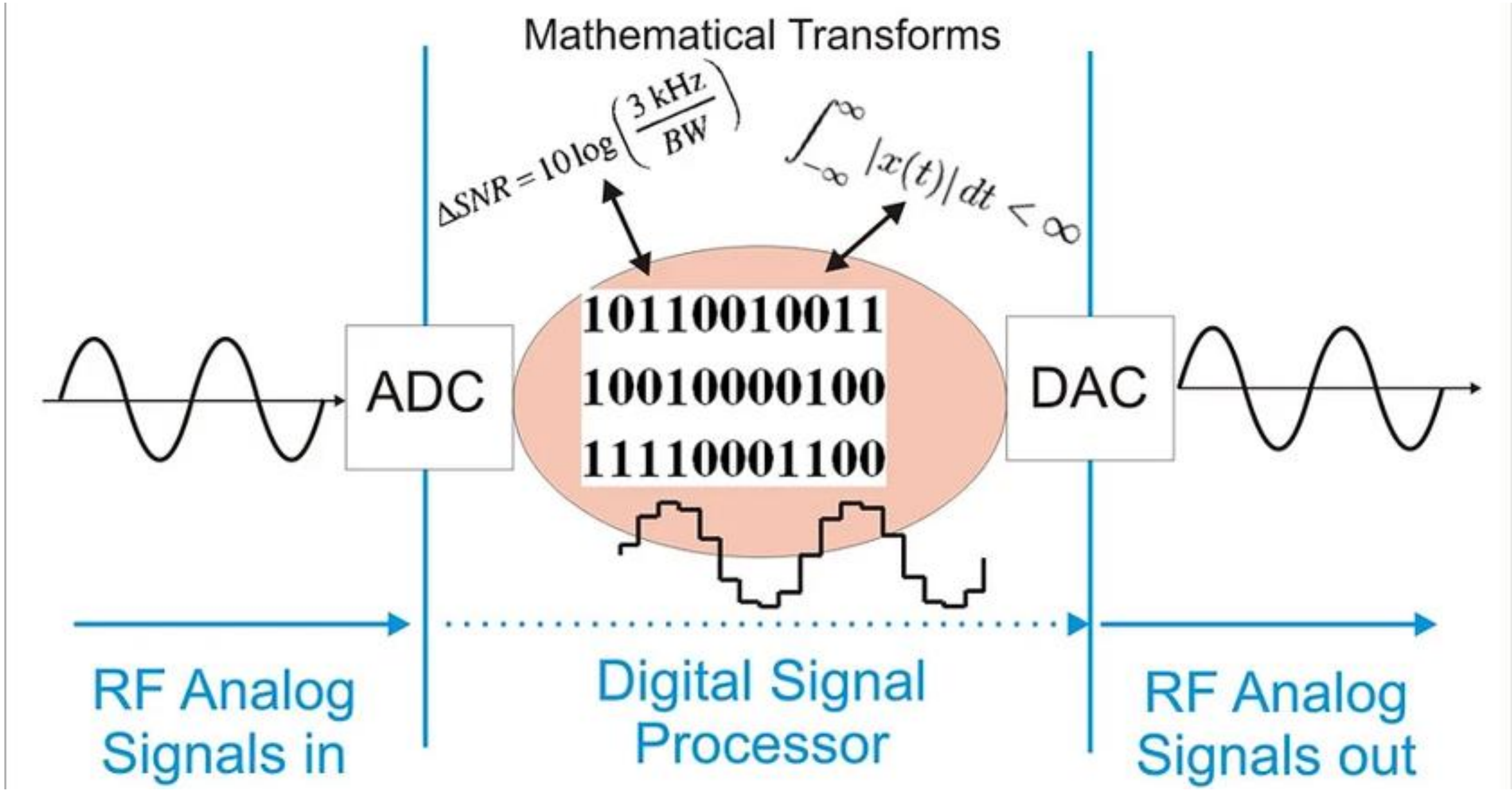
- **WHY?**



VS.

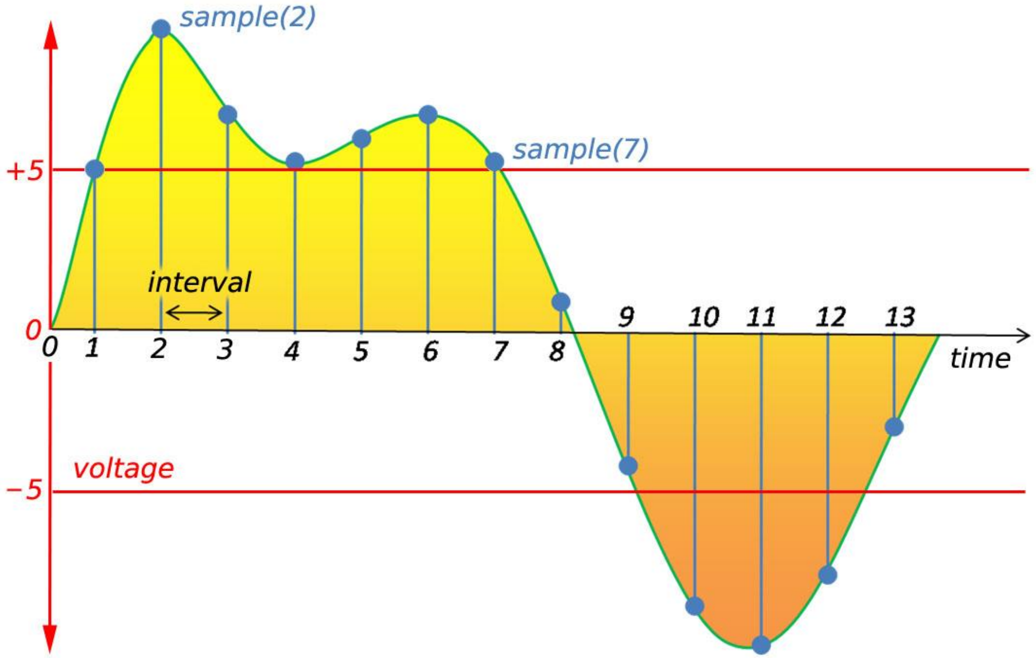


6.3 DSP & SDR


Mathematical Transforms
$\Delta SNR = 10\log\left(\frac{3\text{ kHz}}{BW}\right)$
$\int_{-\infty}^{\infty} |x(t)|\,dt < \infty$
ADC
10110010011
10010000100
11110001100
DAC
RF Analog Signals in
Digital Signal Processor
RF Analog Signals out

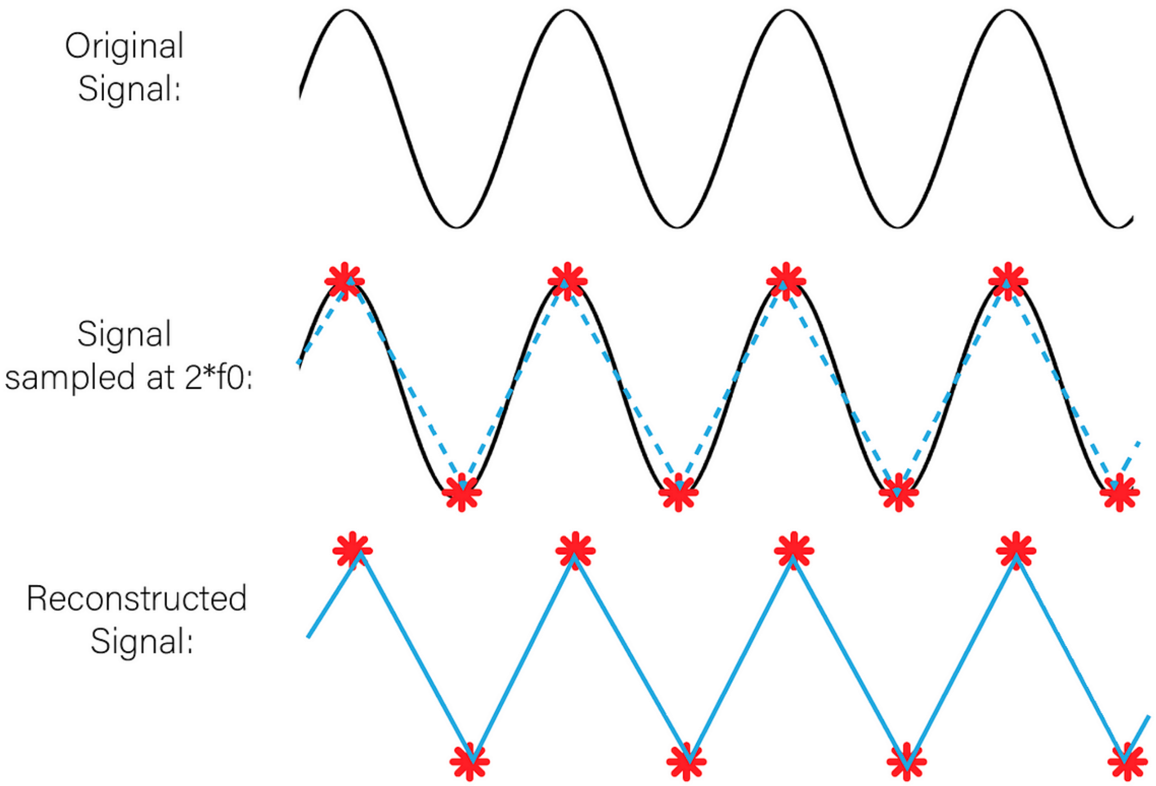
6.3 DSP & SDR

- DIGITAL SIGNAL PROCESSING (DSP)
  - Sequential Sampling
    - Signal sampled at regular intervals
    - The measured amplitude is converted to a digital number



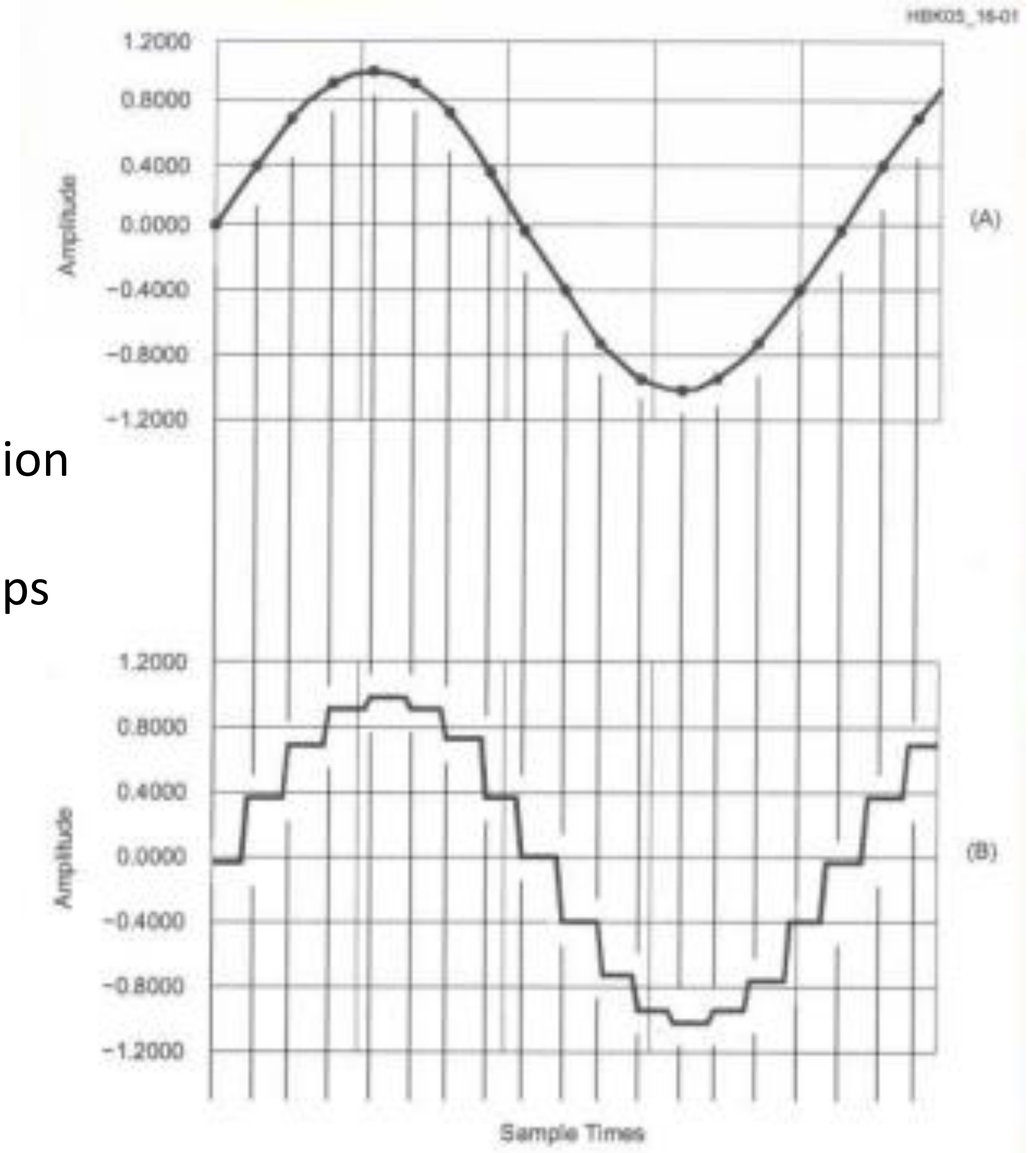
6.3 DSP & SDR

- DIGITAL SIGNAL PROCESSING (DSP)
  - Sequential Sampling
    - Sampling rate must be greater than the frequency being sampled
      - At least twice as many
      - Nyquist Sampling Theorem
    - Sampling Rate determines the TX and RX Bandwidth



6.3 DSP & SDR

- DIGITAL SIGNAL PROCESSING (DSP)
  - Sequential Sampling
    - The more samples, the better reproduction of the original wave.
    - Sampled sine wave is represented as steps
    - Each step adds harmonics



6.3 DSP & SDR

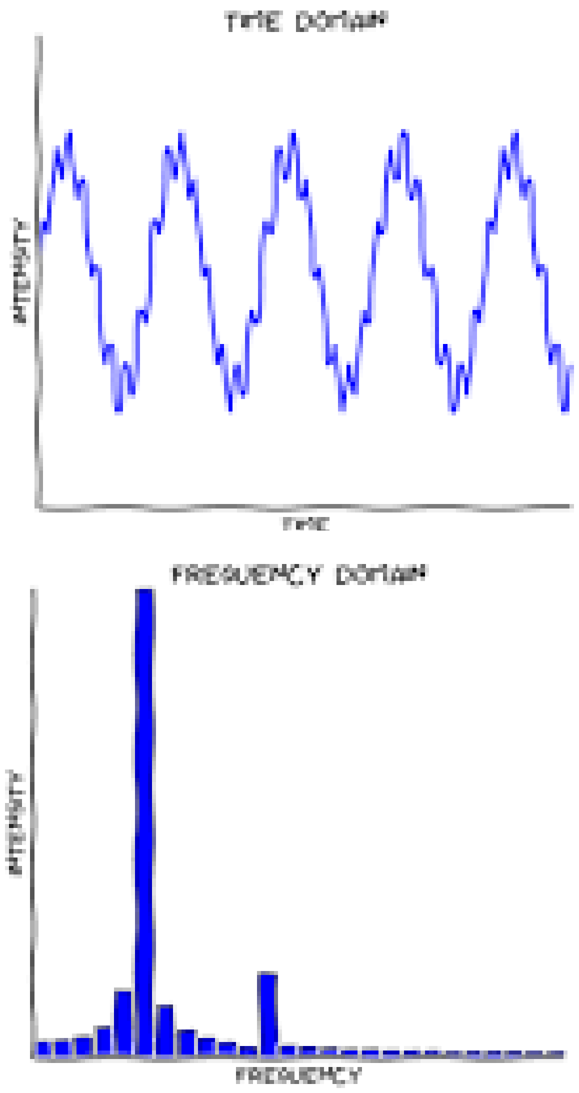
- DATA CONVERTERS
  - Definitions
    - Analog-to-Digital Converter (ADC)
    - Digital-to-Analog Converter (DAC)
  - Samples are measured and the data is converted to a binary number
  - ADC/DAC have limited steps
  - The resolution of the measurement depends on the number of bits
    - 8-bit ADC can record 2^8 values or 256 values
    - Closest value is chosen
    - The more bits, the better the resolution

6.3 DSP & SDR

- DATA CONVERTERS
  - "Dither" – Small amounts of added noise that causes the ADC average values to become more precise over time.
  - ADC resolution determines minimum detectable SDR signal level
  - Converting analog to digital is not exact
  - Total Harmonic Distortion (THD) measures the quality of the conversion

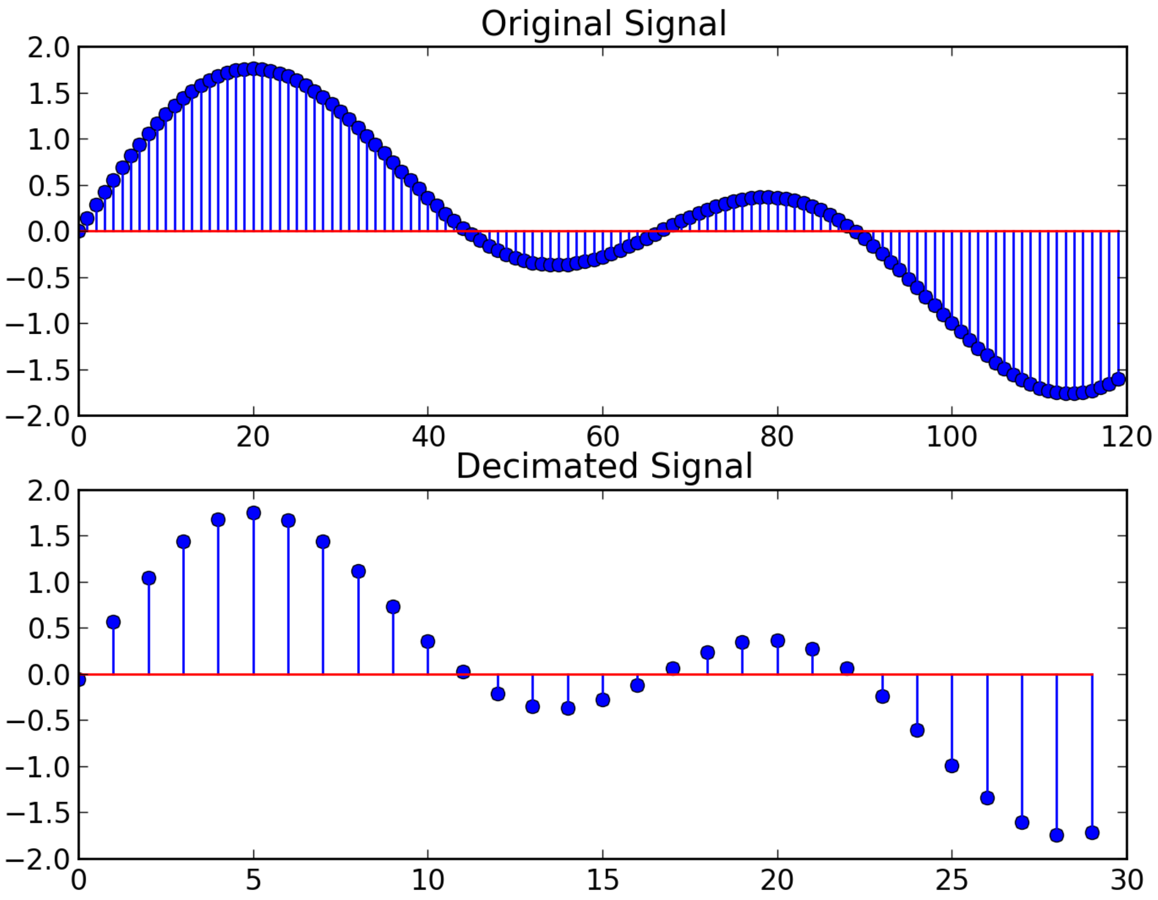
6.3 DSP & SDR

- FOURIER TRANSFORMS
  - Converts signals in the time domain to the frequency domain
  - A signal is comprised of many sine waves combined together.
  - Each frequency comprising the resultant signal gets a bar showing it amplitude
  - Fast Fourier Transform (FFT) is a special algorithm that reduces the number of calculations by a factor of at least 100



6.3 DSP & SDR

- DECIMATION AND INTERPOLATION
  - DSP can do things that can't be done in analog
    - Decimation – removes every nth sample thus reduces the effective sample rate
      - A digital low-pass filter is required to prevent aliasing
    - Interpolation – adds samples in between samples to increase effective sample rate



6.3 DSP & SDR

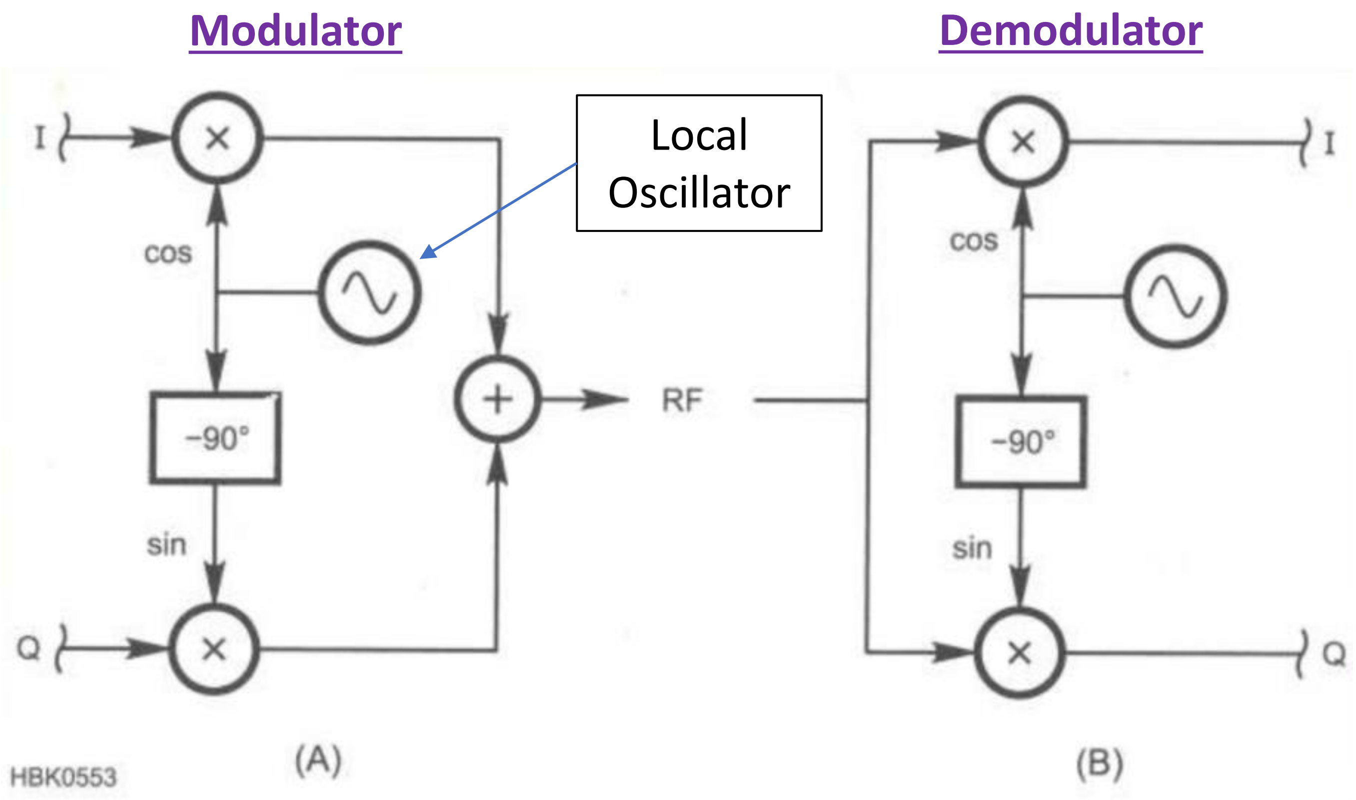
- SOFTWARE-DEFINED RADIO (SDR) SYSTEM
  - Software performs the modulation / demodulation of signals
  - Hardware is not specific to a mode, signal bandwidth or frequency
  - Filtering is performed by software calculation not hardware
  - New functions and features can be added by simply downloading new software

6.3 DSP & SDR

- SDR HARDWARE
  - The conversation from analog-to-digital and vise versa can occur at any point
    - Sound Card @ PC
    - IF stage of the transceiver
    - Direct Digital Conversion (DDC)
      - Very high sampling rates
    - Successive Approximation or Sigma-Delta ADC
    - Only limited by sample rate

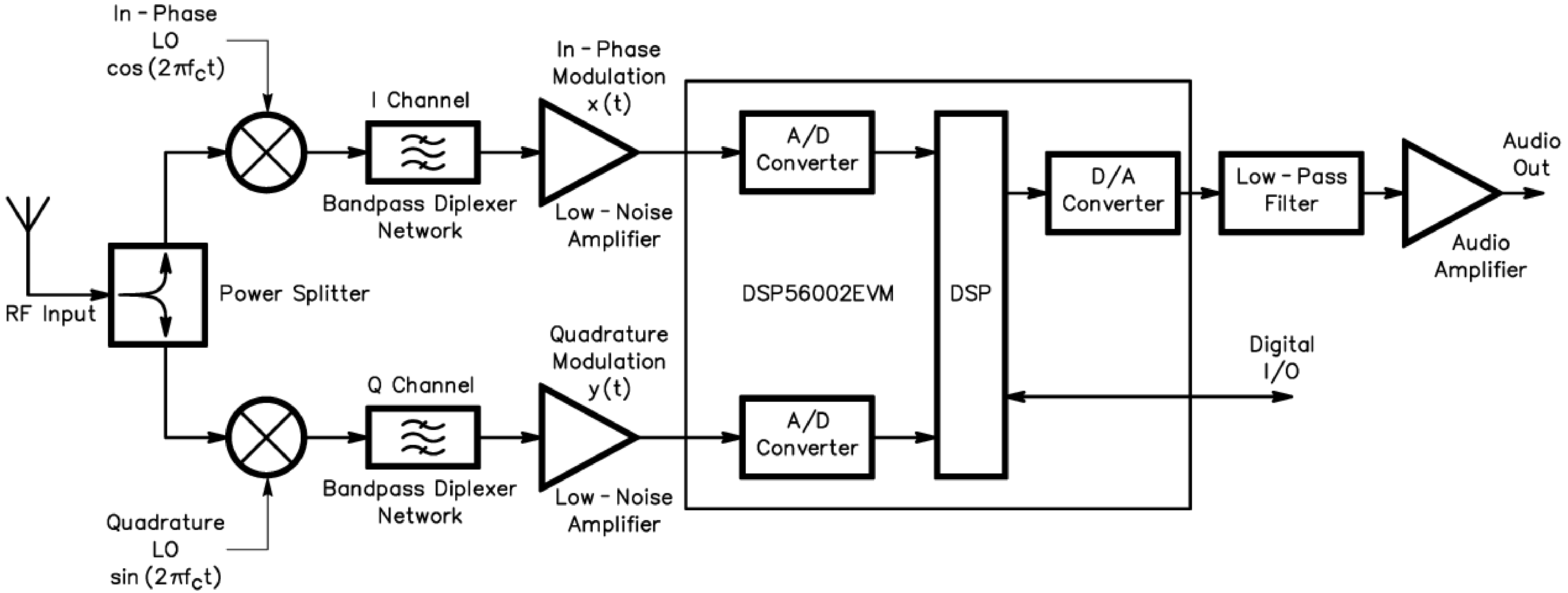
6.3 DSP & SDR

- I/Q MODULATION AND DEMODULATION
  - I & Q define amplitude & phase
  - X = I + jQ
  - Local Oscillator (LO)
  - 90 degrees out of phase
  - Modulation & Demodulation are the same just in reverse



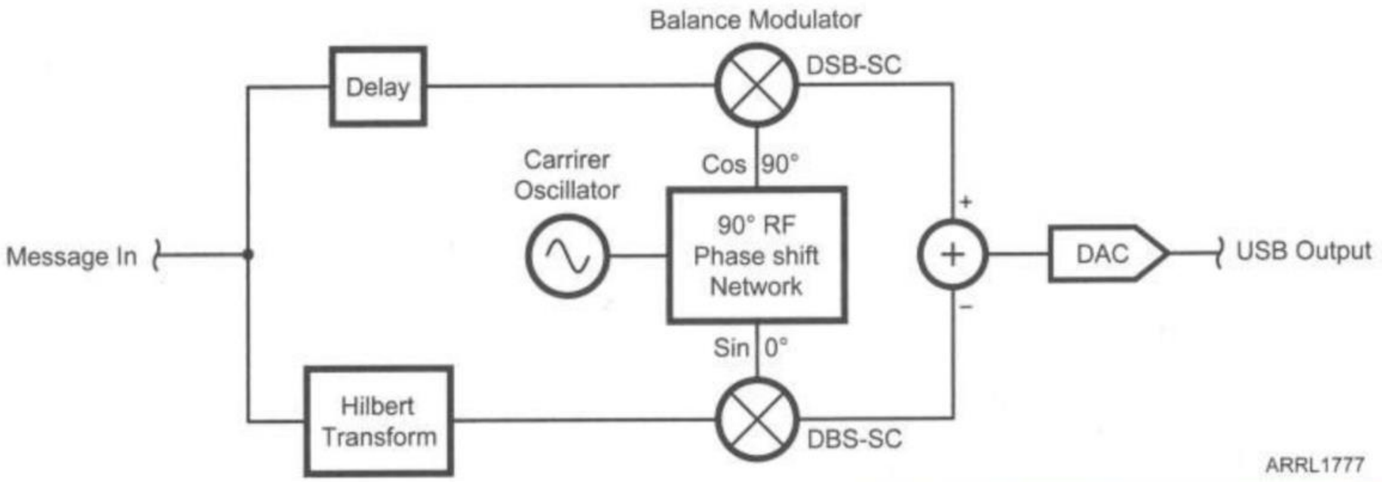
6.3 DSP & SDR

- Direct Digital Conversion



6.3 DSP & SDR

- I/Q MODULATION AND DEMODULATION
  - Common way to use SDR to generate SSB signals
    - Quadrature Phase relationship
    - Hilbert Transform filter
    - Needs corresponding delay



6.3 DSP & SDR

- Any Questions?