Overview

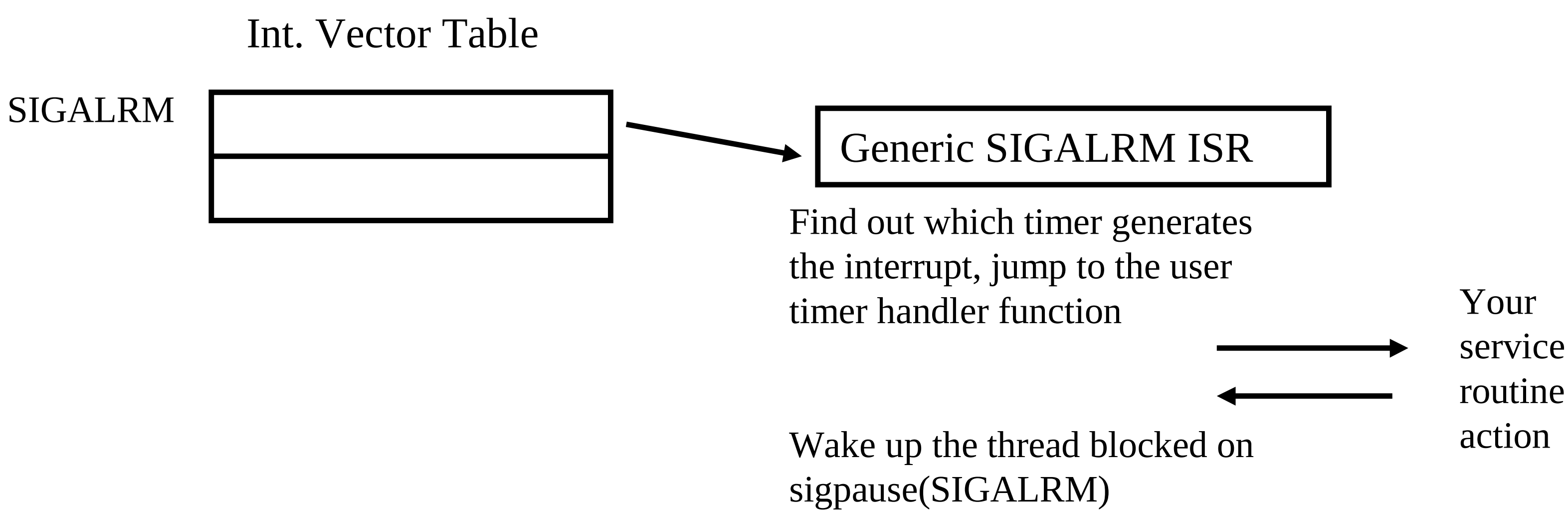
- Signals and data are at the interface of embedded software engineering and ECE.
- Objective:
  - understand the fundamental concepts
  - master issues that a software engineer should do and can do

- Today: signals and data acquisition
  - Source of deterministic errors and random noises
  - Basics of signal spectrums

- Next lecture
  - Review of basic concepts in signal representation
  - Basic filters: what you can do and when you should ask for help

How Does Sigpause Work?

- When you call signal pause, your thread is blocked on a semaphore associated with SIGALGM.
- When your timer handler is done with data I/O part and the control returns to the generic ISR of SIGALGM, the generic handler unlocks the semaphore and wakes up the thread that does the computation part.



POSIX RT Timer Interrupt Handler

- The POSIX timer generates SIGALRM signals (software interrupts). A handler (ISR) for SIGALRM is therefore needed.
- Recall that you first define an interrupt type number which stores the address of your interrupt service routine.
- By POSIX coding convention, you define a generic action and then bind the action to a specific handler your wrote

```
…
struct sigaction my_action, old_action  //actions to catch a given signal arrives
...
// establish signal handler for SIGALRM
memset(&my_action, 0, sizeof(my_action)); // clear up the memory, a convention

my_action.sa_handler = (void *) timer_handler;  //the "action" is to handle  timer interrupt

return_code = sigaction(SIGALRM, &my_action, &old_action); //binding action & signal
```

Deterministic Errors

- Bias adds an offset to the true value. Most analog components have this problem over time.
- Solution: calibration by potentiometer (variable resistor) or by compensating for the bias in software. (Did you find bias in Lab 1's A/D – D/A conversion?)

- Quantization errors: the resolution is not fine enough for the application needs.
  - Proper configuration of voltage ranges in A/D and D/A. (Do you still remember how to do it?)
  - Standard A/D and D/A card gives 12 bit over a given range. However, higher resolution ones also available.

- Aliasing: to be discussed later

Sources of Random Noises

- Noise is the electrical signal that you do not want. In fact, the music broadcast from radio stations is a common source of EMI problems.

- Internal in the electronics of the A/D - D/A card. it is usually very small.
- Environmental :
  - Noise from equipment power transformer. Switching transformers are cheap. The very inexpensive ones are quite noisy. Linear transformers are expensive and heavy, but quiet.
  - Noise from the power line due to power tools, elevators etc. Buy line voltage conditioner or move to another place to do your work.
  - EMI: electrical shielding and/or using differential mode.
  - Do you still remember what is differential mode inputs?

  - The scope is your best friend to identify the source of noise.
  - Poor grounding is a common source of noise problems. Before trying anything fancy, make sure that your equipment is properly grounded.

Ground

- Ground is just like the ground floor of a building. The ground floor of a building on top of a mountain is a lot higher than …

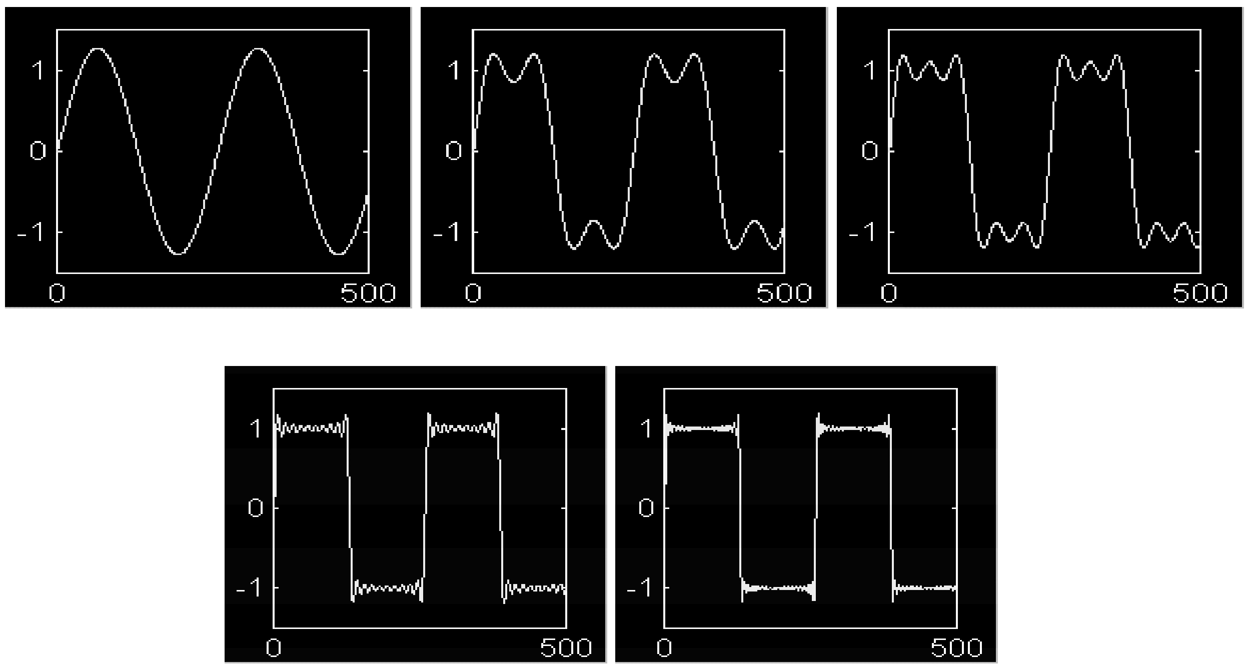
- Electrical ground is the common reference point. Thus, sometime it is also labeled as COMMON. Then there is also the GROUND, the voltage level of the earth. This is like "god" and "God".

- Best Practice
  - connect the signal grounds together and then directly connect it to earth ground via a good GROUND connection.
  - connect the power grounds together and then directly connect it to the earth ground via a separate GROUND connection.

  - Connecting power ground and signal ground together and then run a long wire to the GROUND is not a good idea. The resistance in the ground wire would allow noise in the power equipment to interfere the signal measurement. This is acceptable only if the power equipment is not very noisy.

Fourier serious Pictures

Fourier discovered that a signal can be decomposed into sum of sinusoids. If you have a set of samples, you can use FFT in Matlab to take a look about its frequency components.



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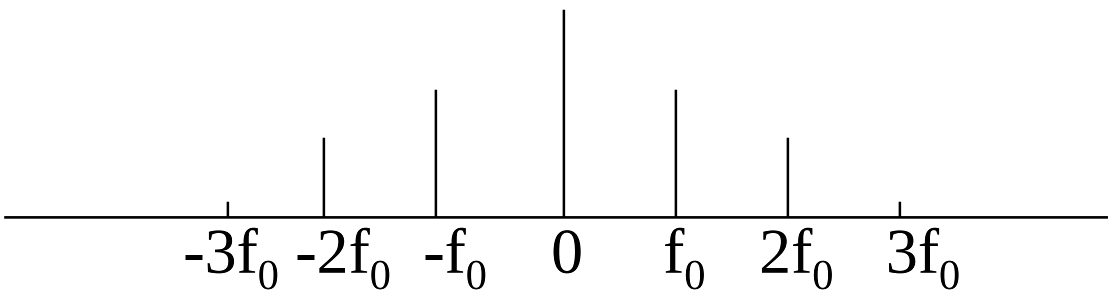
Basic Signal Processing Concepts: Fourier Series

For any periodic function, $f(t)$, whose period is $T = 1/f_0$,

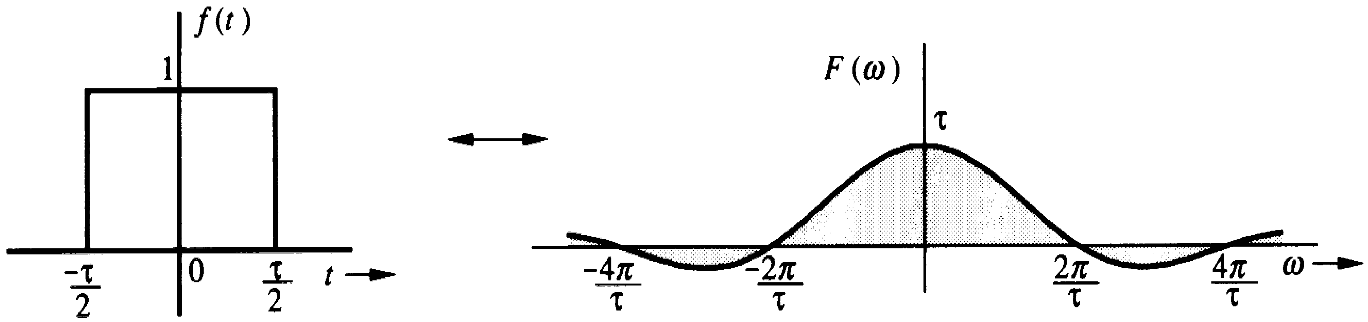
$$f(t) = C_0 + \sum_{k=1}^{\infty} C_k \cos(2\pi f_0 kt + \theta_k)$$



Basic Signal Processing Concepts: Fourier Transform

A signal f(t) is called an energy signal if $\int_{-\infty}^{\infty} | f(t) | dt < \infty$

The Fourier transform of the signal is $f(f) = \int_{-\infty}^{\infty} f(t) e^{-j2\pi t} dt$



Key observation: the narrower the pulse, the wider is the frequency range due to the role played by pulse width in the transformation.

Key Concepts in Signal Processing - 1

- The term *bandwidth* refers to the ranges of frequency components of a signal that has non-negligible magnitudes with respect to the application at hand.

- The most common noises are high frequency noises and thus the popularity of low pass filters.

- Noise in the same frequency ranges as the signal can only be filtered by complex model based filtering, e.g, Kalman filters. If you have significant noises in your signal range, try to prevent them at the source (shielding, differential inputs, better transformers, proper grounding etc).

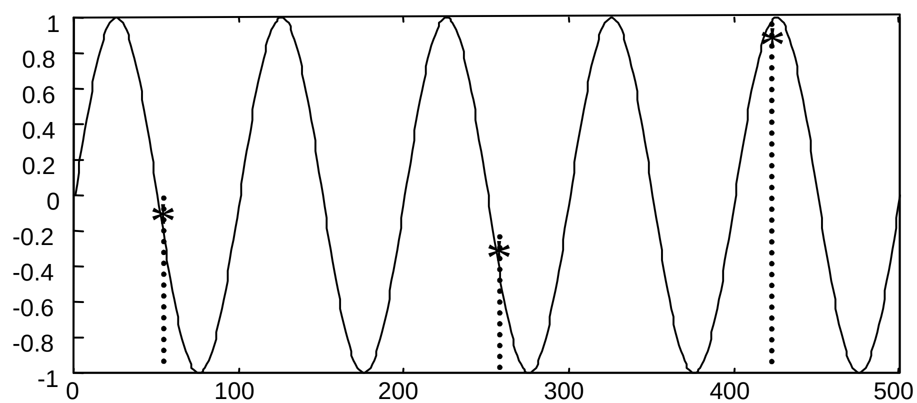
- INCORRECT SAMPLING WILL TRANSFORM NOISE THAT IS OUTSIDE OF SIGNAL FREQUENCY RANGE INTO A NEW NOISE THAT IS IN THE SINGAL FREQUENCY RANGE.

Key Concepts in Sampling

- Nyquist showed that to correctly capture an analogy signal digitally, the sampling rate must be at least twice the bandwidth of the signal.

- This 2 times result assumes perfection in the sampling and filtering process. You won't have it in practice. The practical rule of thumb is at least 3 times.

- Aliasing. If you sample too slow, the high frequency components will become irregular noise at the sampling frequency. The noise can be in the same frequency range as your desired signal!!!



Applications of Nyquist Theorem - 1

What is signal is in the eyes of the beholder. Nyquist was interested in digital samples that capture all the information in the electrical waveform.

The term signal in Nyquist theorem means the total information in the waveform. His signal means the sum of

the “real signal” that you love and
the noise that you hate

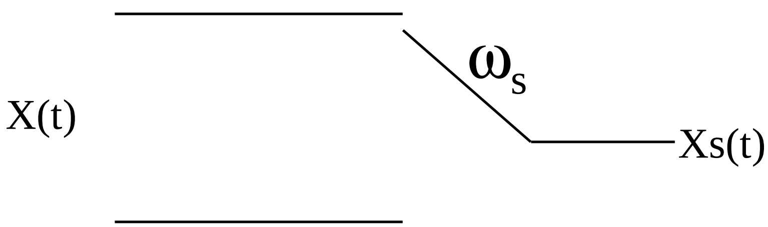
This “love and hate situation” makes the correct application of Nyquist theorem interesting.

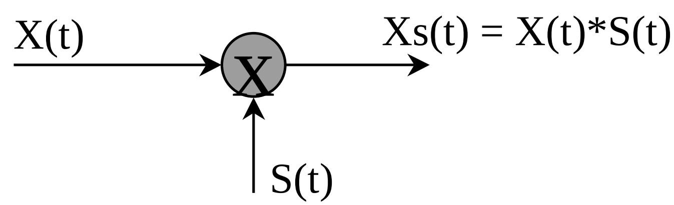
Supposed that we have a signal has frequency components in the range of 10 to 60 Hz and noise ranges from 100 to 500 Hz. What is a minimal and preferred practical sampling frequencies?

Applications of Nyquist Theorem - 2

- If you are sampling below 3*500 Hz, a portion of the 500 Hz noise will be transformed into lower frequency noises that you cannot filter away using classical frequency domain filters.
- 1500 Hz is way too fast for a standard PC with 10 msec clock. You need DSP hardware if you want to digitally filter out the noise after sampling.
- Another way is to use what is known as anti-aliasing filter. A simple hardware analog filter before sampling process begin.

Appendix: A/D Sampling



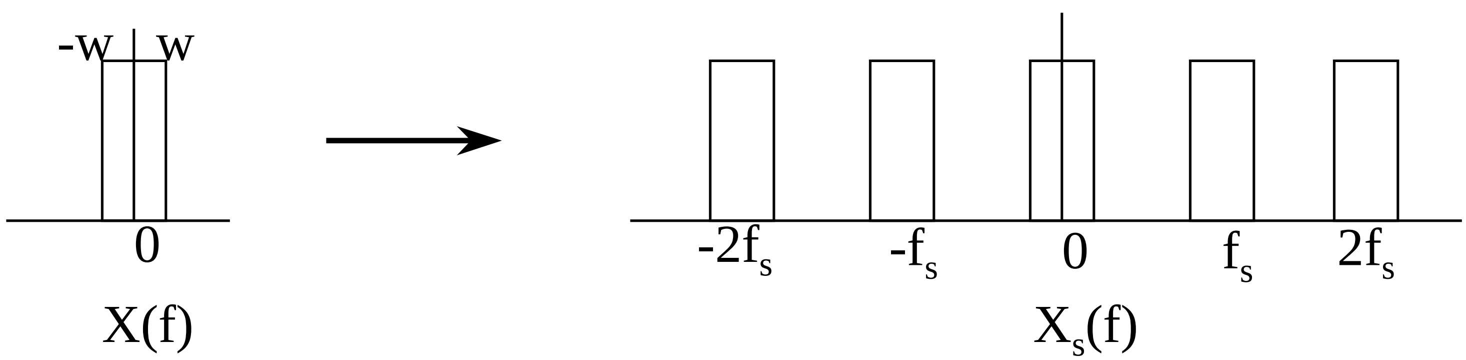


$$S(t) = \frac{\tau}{Ts}\left(1 + \sum_{n=1}^{\infty} 2A_n \cos n\omega_s t\right)$$

S(t) is a rectangular pulse train, where τ is the contact time and ω_s is the sampling rate.

Appendix: Basic Signal Processing Concepts: Modulation Theorem

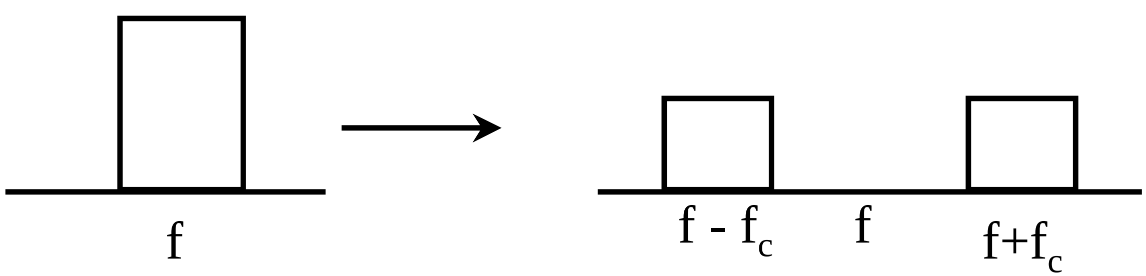
The effect of chopper sampling is to multiply a harmonic series of cosine functions to the signal. By the Modulation Theorem, the cosine functions splits and frequency shifts the signal. The higher the sampling rate, the bigger is the frequency shift and the separation between the shifted signals.



Appendix: Basic Signal Processing Concepts:
Modulation Theorem

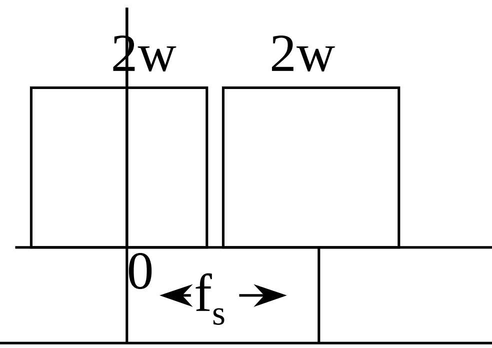
$$v(t)\cos\omega_c t \longrightarrow 1/2\ (V(f - f_c) + V(f + f_c))$$

Multiply a signal *v(t)* by a cosine function with frequency f_c splits the signal into two parts and frequency shifted by f_c and $-f_c$



Appendix: Basic Signal Processing Concepts:
Nyquist Rate and Aliasing

For a signal with bandwidth W, the amount of frequency shift, f_s, must be greater than 2W or there will be overlaps between the spectrums of the shifted signals. The overlap, if any, is called aliasing.



It follows that the sampling rate must be at a least two times of the bandwidth, the highest frequency in the signal, to avoid aliasing. This is known as the Nyquist rate. If there is no aliasing, the the signal can be recovered perfectly in theory by ideal low pass filters.