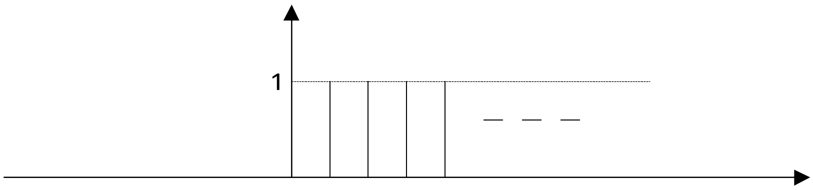
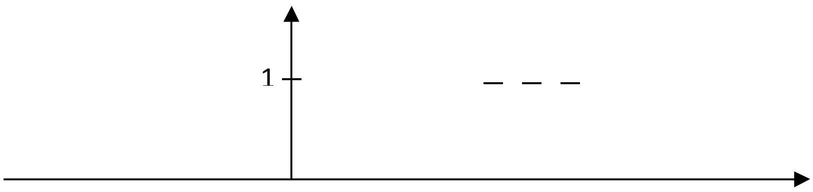
Revision of DSP signals

There are a number of important basic digital signals used in DSP analysis. These include:

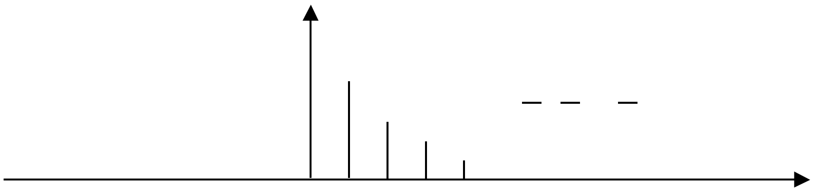
Unit Step: Defined as u(n)=1, n≥0 and zero elsewhere.



Unit Impulse: defined as δ(n)=1, n=0 and zero elsewhere.



Decaying Exponential: defined as g(n)=a^n, n≥0 and zero elsewhere where 0<a<1



Digital sequences

A digital sequence is defined as a sum of scaled delayed impulses.

$$x(n) = a_0\delta(n) + a_1\delta(n-1) + a_2\delta(n-2)$$

$$In\ general\ x(n) = \sum_{k=0}^{N} a_k\delta(n-k)$$

Problem 1:

Write down the digital sequence for *x(n)* if *N=4* and a_k*=10,8,4,5,2* for *k=0 to 4*. Draw a sketch of this sequence.

Problem 2:

The digital sequence in Problem 1 is added to the following sequence:

$$y(n) = \sum_{k=0}^{6} a_k\delta(n-k)\ ,\ a_k = 2b^{k+1}$$

Sketch the combined sequence if:

(i) b=0.5
(ii) b=2

Sampling & Digital Frequency

Nyquist rate

All signals must be sampled at a least twice their maximum frequency. This sampling rate is known as the nyquist rate.

Discrete Sinusoidal Signal

In discrete form the tone is defined as:

$$x(n) = A\sin(\omega_a nT), T = \frac{1}{f_s}, f_s = sampling\ freq$$

For a sampled signal time '*t*' is replaced with *t=nT* where *T* is the sampling period.

$$x(n) = A\sin(2\pi n f_a / f_s)$$

The factor $2\pi\left(\frac{f_a}{f_s}\right) = \theta, the\ digital\ frequency$

It represents the angle that the analogue signal rotates through during one sampling period.

$$x(n) = A\sin(n\theta)$$

If for example the analogue signal is a single tone of frequency 1 kHz and the sampling frequency is 8kHz, then $\theta = 2\pi \times 1/8 = \pi/4$.

Spectral Analysis

We can obtain the frequencies in a signal by using the Fourier Transform. We can use the FFT (Fast Fourier Transform) in the Pspice or Matlab environment.

A piece of Matlab code is shown below for obtaining the spectrum of a signal:

```
delfreq=fs/N;                      % Frequency resolution
freqval=(1:N/2);                   % Symmetrical about fs/2
freqval=freqval*delfreq;           % Frequency axis
spect=abs(fft(hamming(N)'.*tone)); % abs produces magnitude spectrum from fft
spect=spect(1:N/2);
```

If we sample a signal for N samples, then the frequency resolution is fs/N where fs is the sampling frequency. Only half of the values in the spectrum contain useful information since the spectrum repeats or fold-overs at half the sampling frequency. It is often common practice to 'window' the signal to improve the spectral response.