

12.444 Optical Components and *Communication*

(Communications part by Klaas Wynne Components part by Ivan Ruddock)

Content of Lectures in 1998:

- **Tuesday, September 29, 1998:** ~40min., general introduction, **Analogue Communication**, general communication system, AM vs. FM, fields vs. powers, Fourier transforms, sampling theorem.
 - **Thursday, October 1, 1998:** 55 min., **Digital Communication**, digitisation, pulse code modulation, time division multiplexing, noise, inter symbol interference, bit error rate, relation to signal-to-noise ratio, sources of noise.
 - **Tuesday, October 6, 1998:** 55 min., Poisson statistics, information theory, **Fibre Systems** (intro.).
 - **Thursday, October 8, 1998:** **Fibre Systems**, dBm, power budget (viability), detector noise, photodiodes, APDs, PMTs, MCPs, photons.
 - **Tuesday, October 13, 1998:** **Amplifiers**, optoelectronic repeaters, EDFAs, **Practical Communication Systems**, time-division multiplexing, MOSFETs, ring counter, buffer delays, network, long-haul telecommunications.
 - **Tuesday, October 20, 1998:** INTERNET, packet switching, topology, chaotic packet traffic, **Coherent Detection**, heterodyning.
 - **Tuesday, October 27, 1998:** High bit rates and material **dispersion** (phase velocity, group velocity, group velocity dispersion).
 - **Tuesday, November 3, 1998:** **The speed of light and signals.**
 - **Tuesday, November 10, 1998:** Ill.
 - **Tuesday, November 17, 1998:** Broadening of pulses in fibres due to GVD. Dense Wavelength Division Multiplexing (DWDM).
 - **Tuesday, November 24, 1998:** **Solitons.**
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0 Introduction

0.1 Overview of the Communications Class

Class Details: Semester I, 1 credit, 24 lectures, 6 tutorials

Textbook: Essentials of Optoelectronics, A. Rogers (Chapman & Hall)

Learning Outcomes: On completion of the class, students should understand the principle of operation of optical devices used in light manipulation and optoelectronics and have a good understanding of optical communication systems. Students should be able to show that they understand the following optics and solve related problems.

Summary of Lectures:

Planar dielectric waveguides (ISR): Revision of Fresnel equations, total internal reflection and phase shifts. Planar dielectric waveguides and modal resonance condition for asymmetric guides in detail. Minimum guide thickness and "V" parameter. Modal patterns for symmetric guides. Fabrication and coupling. Integrated optical (IO) phase shifter and Mach-Zehnder interferometer.

Optical fibres (ISR): Step index fibres, numerical aperture and acceptance angles. Step index modes qualitatively - TE, TM, HE, EH, LP. Mode sequence and "V" parameter. Intermodal, waveguide and group velocity dispersion. Microbending, scattering and absorption losses. Birefringence in single mode fibre. Low and high birefringence fibres. Graded index fibre.

Optical fibre sensors, coherent and incoherent, passive and active. Optical fibre sensors of position, pressure, temperature, magnetic field and liquids in varying detail.

Basics of communications (KW): The generalised communication system including amplitude and frequency modulation, sidebands and pulse code modulation. Fourier-transforms and the sampling theorem.

Digital communication (KW): Pulse code modulation in detail. Noise sources, definition of the bit error rate, signal-to-noise ratio, bandwidth and intersymbol interference. Introduction to information theory,

information, entropy, channel capacity and the ASCII system. Information in a two-dimensional picture and in written text.

Fibre systems (KW): Design principles of optical fibre systems. Attenuation and power budgets. Splice and connector losses. Detectors: photomultiplier, pin photodiode, and avalanche photodiode. NEP and D^* , sources of noise, Johnson and shot noise, quantum noise. Fibre amplifiers and lasers.

Communication systems (KW): The telephone system, time division multiplexing, exchanges, long-haul telecommunication systems. The Internet.

Recent developments (KW): Coherent optical communications. High bit rates and material dispersion (phase velocity, group velocity, group velocity dispersion). Wavelength division multiplexing (WDM). Communication using optical solitons.

Assessment: 1½ hour examination in January (2 out of 3 questions, no resit).

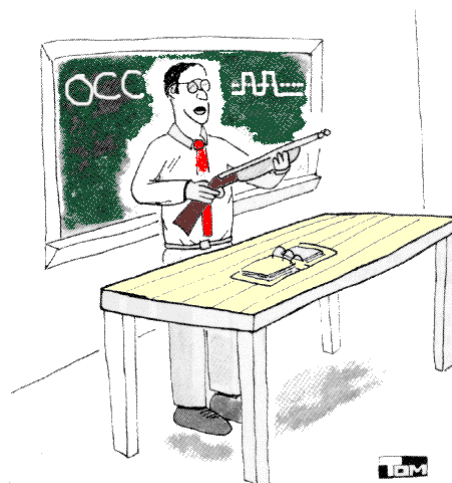
0.2 Some books

- "Essentials of Optoelectronics" by A. Rogers, Chapman & Hall, 1997. The official textbook of the course. Rather heavy on the components side, rather light on the communications side.
- "Introduction to Communication Systems" by F.G. Stremler, Addison-Wesley, 1982. Some good bits about sampling theorem, modulation, circuits, etc.
- "Optoelectronics. An Introduction" by J. Wilson, J.F.B. Hawkes, Prentice Hall, 1983. About lasers, fibres, detectors, etc.
- "Telecommunications Networks" by F. Mazda, Focal Press, 1996. About networks, telecommunications standards, etc.
- "Fibre Optic Systems" by P. Halley, Wiley, 1987. The title says it all.
- "Optical Fiber Communications. Principles and Practice" by J.M. Senior. Idem.
- "Telecommunications Engineering" by J. Dunlop, D.G. Smith, Chapman & Hall, 1995.
- "Optical Electronics" by Yariv, Harcourt, 1991. Yariv has written several very similar books. They are good for nonlinear optics, physics of light, etc.
- "Science and Information Theory" by L. Brillouin, Academic Press, 1962. Great book to read during your Christmas holiday and learn some real physics.
- "The Physics of Information Technology" by N. Gershenfeld, Cambridge, 2000. Just got this book, so can't give a good opinion but it looks very promising.
- "Numerical Recipes in C" by W.H. Press, S.A. Teukolsky, W.T. Vetterling, B.P. Flannery, Cambridge, 1995. The chapter on FFTs explains the sampling theorem very well.

0.3 Who is this guy?

My name is Klaas Wynne, I can be found in room JA8.14. Telephone: (141) 548-3381, e-mail: klaas.wynne@phys.strath.ac.uk, URL <http://dutch.phys.strath.ac.uk/FRC/>.

"PLEASE FEEL FREE TO INTERRUPT
IF YOU HAVE A QUESTION."



1 Communication Systems

The generalised communication system including amplitude and frequency modulation, sidebands and pulse code modulation. The sampling theorem and its proof.

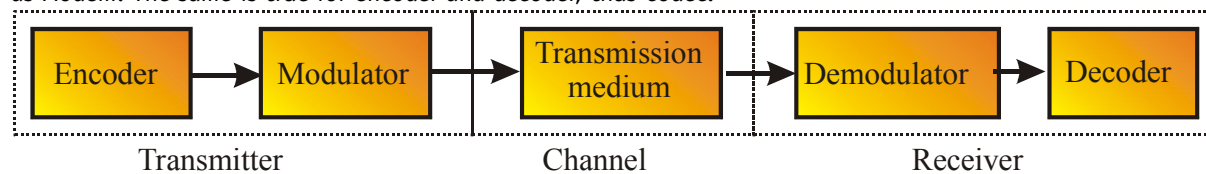
1.1 Introduction

History Beacon fire / smoke signals / etc.

Communication. Satellite, TV, telephone. CNN reports on the Gulf war. Internet. Bank transactions (ATMs). Nothing is more important to business, national welfare and defence than communication these days.

1.2 A general communication system.

A general communication system consists of three parts: (1) The **transmitter**, (2) the **channel** and (3) the **receiver**. The modulator and the demodulator are often the same bits of electronics and tend to be referred to as Modem. The same is true for encoder and decoder, thus Codec.



Transmission can be done: **Simplex**, **half duplex** and **full duplex**. Simplex means that transmission of information is only in one direction (transmitter → receiver). Half-duplex means that information can travel in both directions but there is only **one** channel. Full duplex means that both sides have both a transmitter and receiver, and information is sent both ways through two separate channels.

Examples of transmission media are:

- Electromagnetic waves:
 - telegraph – wire
 - very long wavelength radio waves – space
 - microwaves – free space or cable
- optical wavelength – free space or fibre
- Sound waves
- Carrier pigeon, smoke signals, air mail

The transmission medium introduces distortion, noise and interference. It is typically the limiting factor in any communication system.

1.3 Signals: AM vs. FM

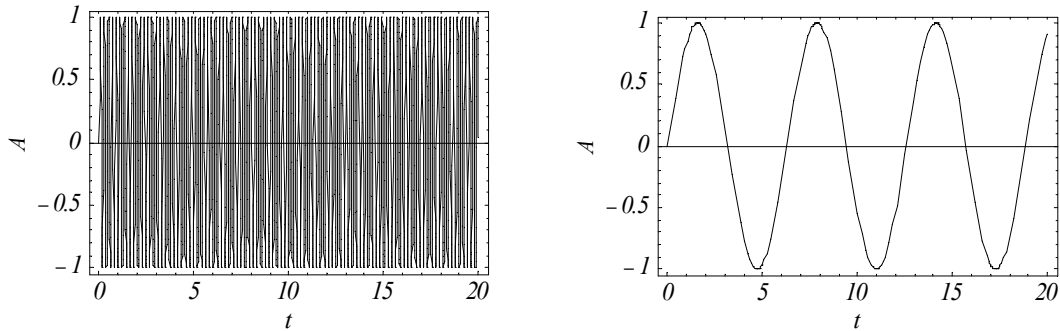
The equation of a general sinusoidal signal can be written as:

$$e(t) = a(t) \cos[\omega(t)t + \varphi(t)] \quad (1.1)$$

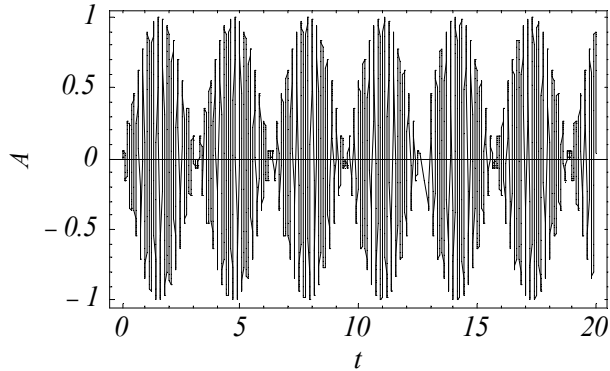
The cosine term at frequency ω , represents the **carrier wave**. Given this general form of a wave, there are now three possible ways to modulate the carrier wave in order to encode information:

- Amplitude modulation (AM) → modulation of $a(t)$.
- Frequency modulation (FM) → modulation of $\omega(t)$
- Phase modulation (PM) → modulation of $\varphi(t)$

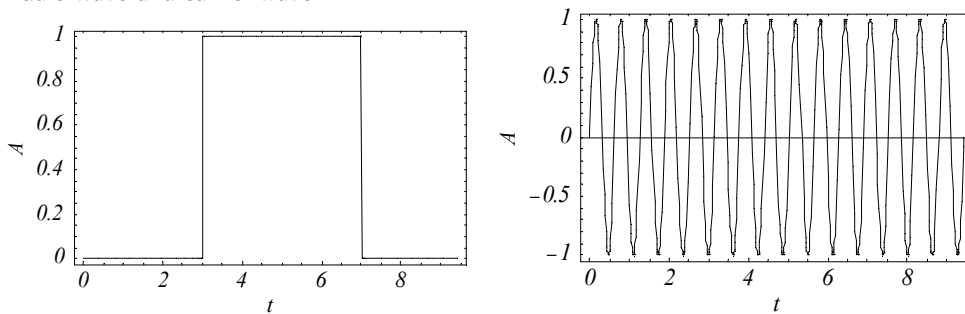
Amplitude modulation (AM). Low frequency information (e.g., audio frequencies) imposed on high frequency carrier wave. For example, the following carrier wave (left) and audio wave (right):



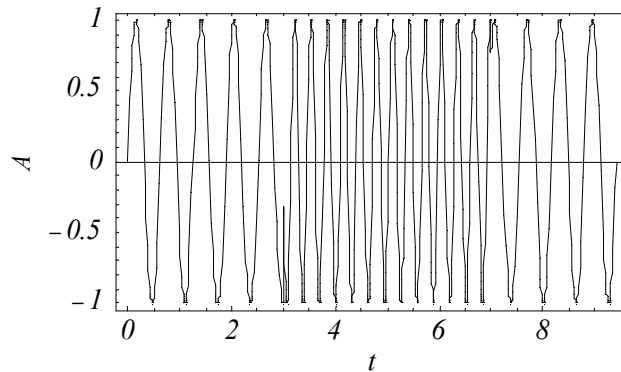
results in the following AM modulated wave:



Frequency modulation. Same basic idea as AM, except now the frequency is modulated rather than the amplitude. Audio wave and carrier wave



Frequency modulated wave:



1.4 Energy Signals, Power Signals

For the systems we will consider here, a signal is a voltage or a current. A typical example is an electrical signal travelling through a copper cable or a microwave signal travelling through the atmosphere. The instantaneous power dissipated by a voltage $e(t)$ in a resistance R is

$$p = |e(t)|^2 / R \tag{1.2}$$

and for a current $i(t)$

$$p = |i(t)|^2 R \quad (1.3)$$

both with units Watts. The power density of a light beam is given by:

$$P = \frac{1}{2} \epsilon_0 c |E|^2 \quad (1.4)$$

where ϵ_0 is the dielectric constant, c the speed of light and E the electric field (power density has units of W/m^2). So ignoring trivial details, the power of a signal is the square of the signal.

Engineers like to use the unit of decibel (dB), which is proportional to a ratio between two powers. For example, the ratio of a signal before it goes into an amplifier vs. when it comes out. The definition of the dB is:

$$20 \log_{10} \left(\frac{A}{B} \right) \quad (1.5)$$

if A and B are **signals** (voltage or current) and

$$10 \log_{10} \left(\frac{A}{B} \right) \quad (1.6)$$

if A and B are **powers**.

1.5 Fourier transformation

The Fourier transformation of a function $f(t)$ is defined as:

$$\tilde{f}(\omega) = \int_{-\infty}^{\infty} dt f(t) e^{-i\omega t} \quad (1.7)$$

and its inverse:

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} d\omega \tilde{f}(\omega) e^{+i\omega t} \quad (1.8)$$

or alternatively the Fourier series. There is a caveat: Signals usually do not last infinitely long. Therefore the FT only applies to pulsed signals or signals that are periodic.

Tables of rules for FT's can be found anywhere. One rule is quite important here, it is the convolution theorem.

$$FT(f(t)g(t)) = \frac{1}{2\pi} \int_{-\infty}^{\infty} dx \tilde{f}(x) \tilde{g}(\omega - x) \quad (1.9)$$

which is of course very useful to understand AM. There is no comparable theorem for dealing with FM. Consider an AM wave:

$$e(t) = a(t) \cos[\omega_0 t + \varphi] \quad (1.10)$$

The Fourier transform of a cosine is:

$$\int_{-\infty}^{\infty} \cos[\omega_0 t] e^{-i\omega t} dt = \pi \{ \delta(\omega - \omega_0) + \delta(\omega + \omega_0) \} \quad (1.11)$$

that is, two Dirac delta functions peaking at plus and minus the carrier frequency. Because of these delta functions, the convolution integral is easy to do and it is found for the FT of the AM wave:

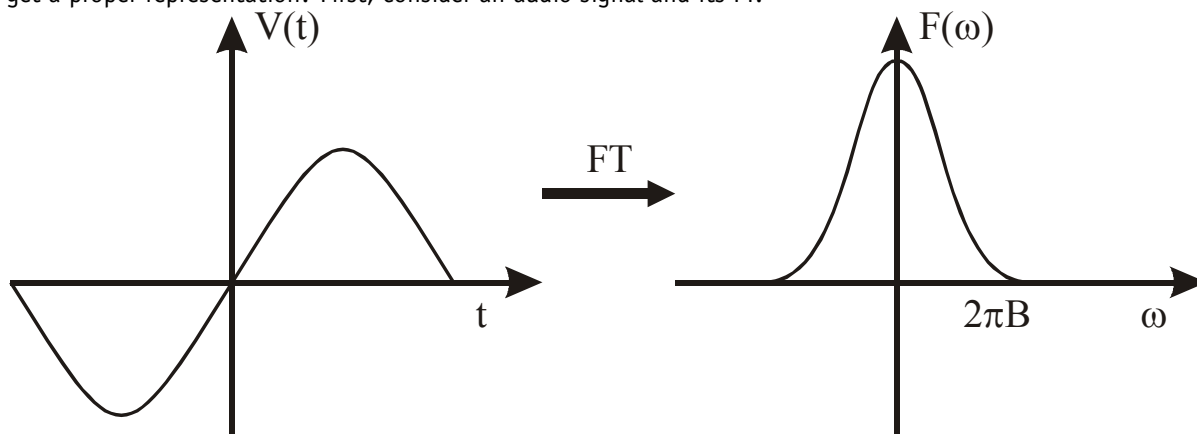
$$\tilde{e}(\omega) = \frac{1}{2} \tilde{a}(\omega + \omega_0) + \frac{1}{2} \tilde{a}(\omega - \omega_0) \quad (1.12)$$

that is, the carrier shifts the audio modulation to higher frequency ($\pm\omega_0$) but otherwise there is no change.

1.6 Sampling theorem

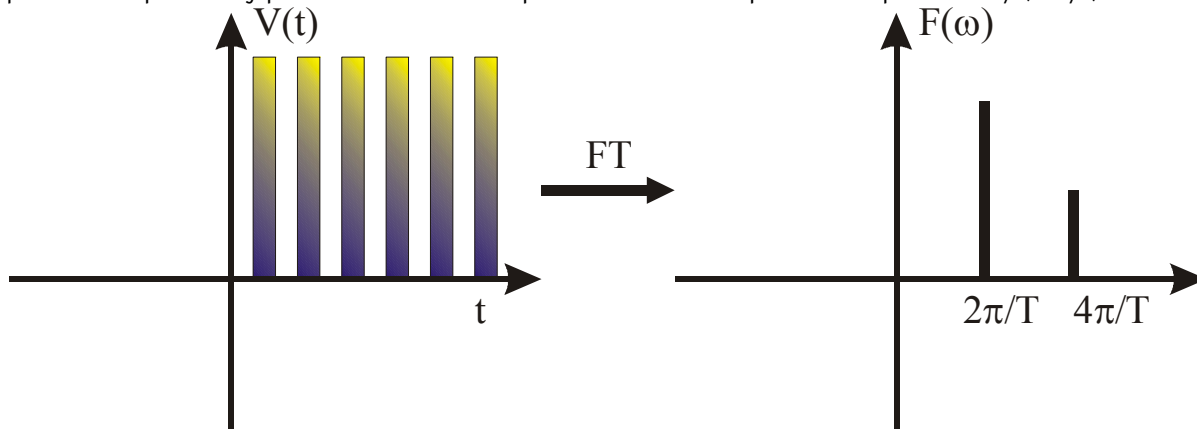
When sending a signal (say audio waves) through a channel, clearly one would like to minimise the amount of required information. Later I will discuss digital communication and in that case, the wave has to be sampled. Each sample is converted to a certain number of bits. All these bits have to be transmitted and a smaller

number of bits means less congestion on the networks. How often does an audio wave have to be sampled to get a proper representation? First, consider an audio signal and its FT:

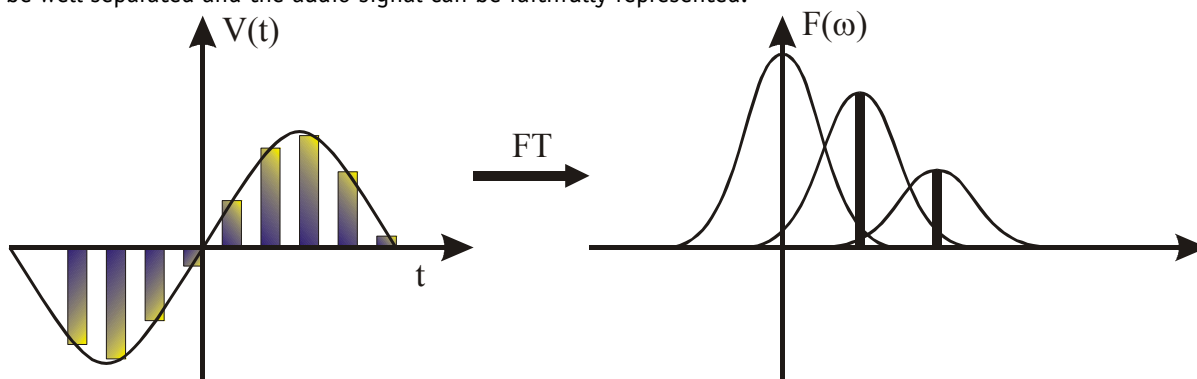


In the picture above, it is taken that the signal has no frequency components above frequency B . As an example, take human speech: Hearing cuts off at 20,000 Hz so that is an absolute upper limit for the frequency. However, in speech most frequencies will be below $\sim 4,000$ Hz. Thus, for applications such as the telephone system, $B = 4,000$ Hz can be taken as the cut-off and for CD-quality sound, $B = 20,000$ Hz.

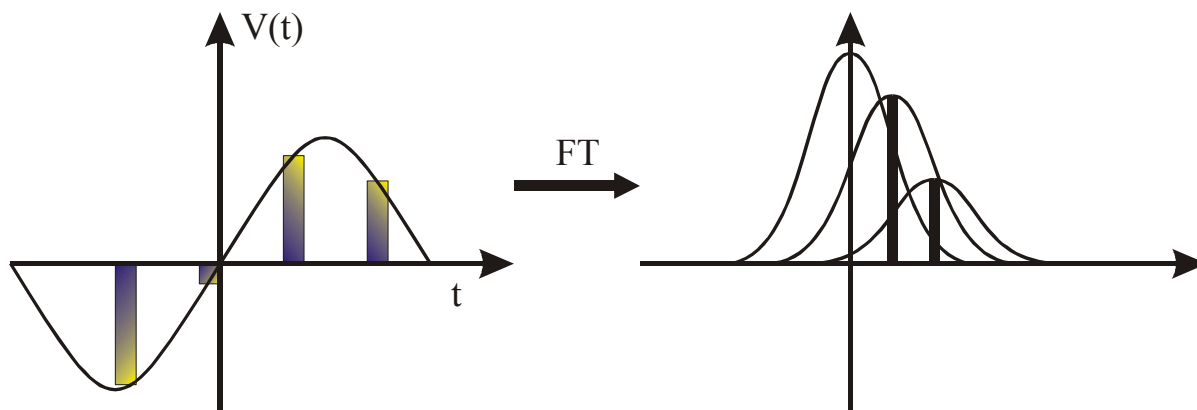
(Warning: The following explanation is fairly crap.) Now consider a train of sampling pulses, where the pulses are separated by period T . The FT of this pulse train will have peaks at frequencies $2\pi/T, 4\pi/T$, etc.:



Now consider the same sampling pulse train modulated by the audio signal. Its FT will consist of a series of peaks at $2\pi/T, 4\pi/T$, etc., where each peak has the same shape as the peak in the FT of the audio signal only. That is, the peaks have a width on the order of $2\pi B$. If the **sampling period is short ($T < 1/2B$)** the peaks will be well separated and the audio signal can be faithfully represented:



If the **sampling period is long ($T > 1/2B$)** the peaks start to overlap and information about the audio signal is lost:

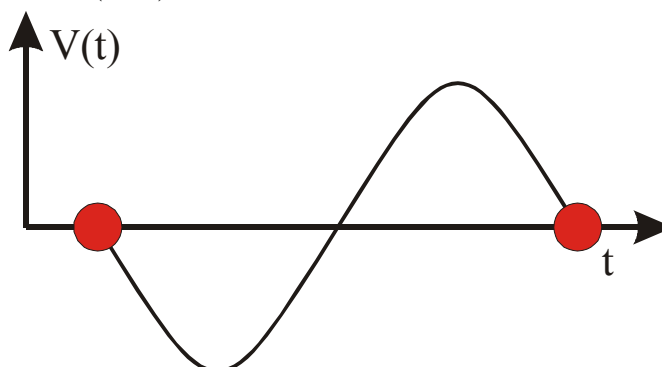


Spacing between peaks: $2\pi/T$. HWHM of peaks: $2\pi B$

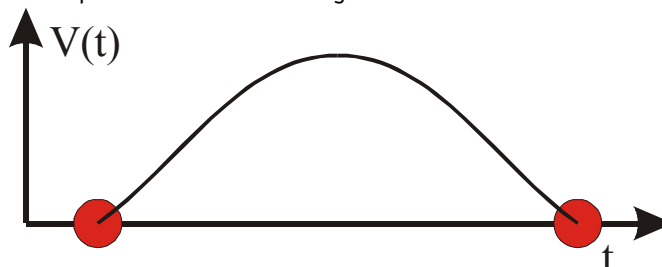
If the different peaks in the Fourier spectrum do not overlap, a low pass filter can be used to recover the central peak, i.e., the frequency spectrum of the original signal. Here $2 \cdot 2\pi B < 2\pi/T$. $T < 1/(2B)$...result of the sampling theorem.

The sampling theorem. Given that we have a time varying signal, how frequently must it be sampled to obtain all the necessary information about the signal? The sampling theorem states that: **A real-valued band-limited signal having no spectral components above a frequency B Hz is determined uniquely by its values at uniform intervals spaced no greater than $1/(2B)$ seconds apart.**

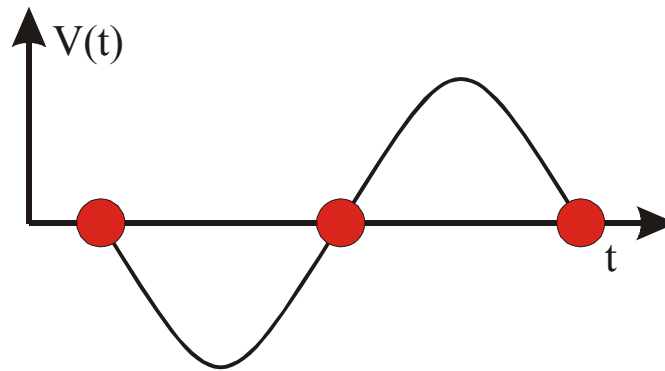
There is an easier way to understand the sampling theorem. A signal wave, for example speech, will contain many frequencies and the signal waveform can look quite complicated. In the case of speech, it will contain frequencies ranging from 0 Hz to 4,000 Hz. Now consider a small section of the signal that happens to be at the cut-off frequency B, i.e., $\sin(2\pi Bt)$:



In this figure, the signal is sampled twice. Is this enough? No! A wave at half the frequency:



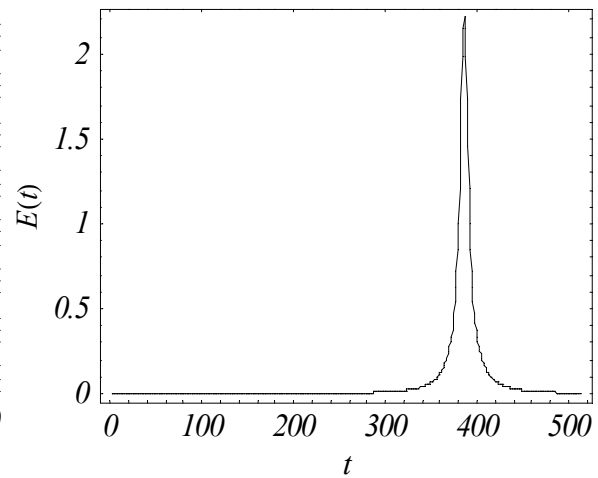
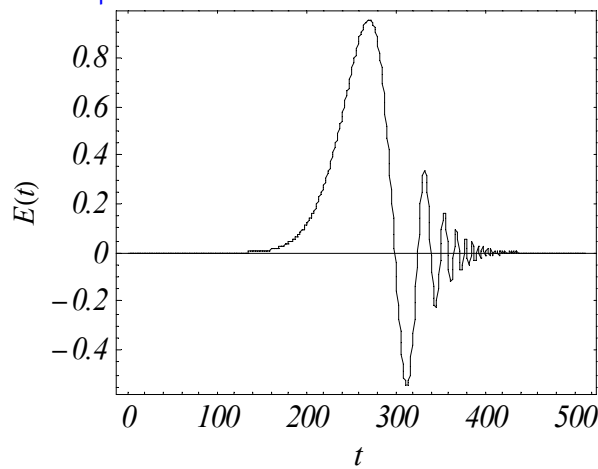
will go through the two sample point as well. Thus, the conclusion is that three samples, for example,



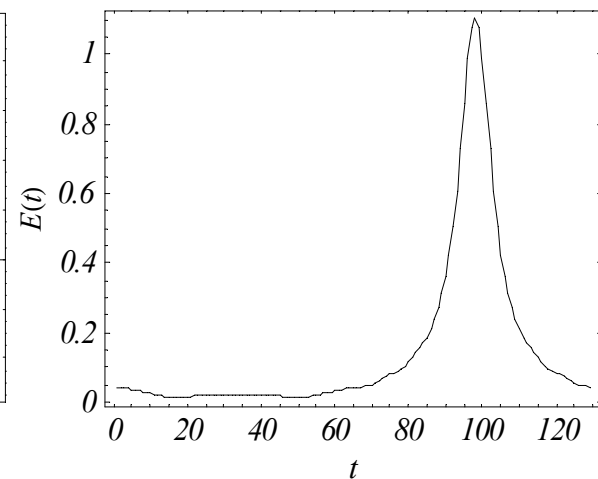
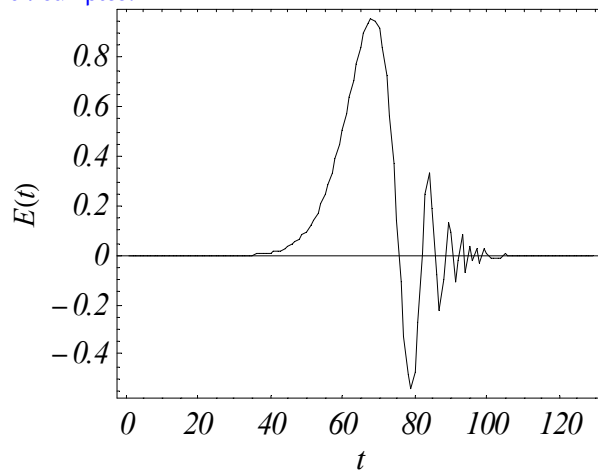
are needed to define a signal properly. With a bit of hand waving, you can therefore understand that to sample a wave at frequency B properly, the sampling frequency has to be $2B$.

To sample a wave at frequency B properly, the sampling frequency has to be $2B$.

256 samples:



64 samples:



16 samples:

