AV312 - Lecture 11

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Figures from "Communication Systems" by Haykin and "An Intro. to Analog and Digital Commn." by Haykin and Moher

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Review of last classes

- Analog modulation and demodulation
  - Amplitude modulation - LC/plain AM, DSBSC, SSB, VSB
  - Frequency modulation
  - PLL
  - Continuous wave modulation for passband channels

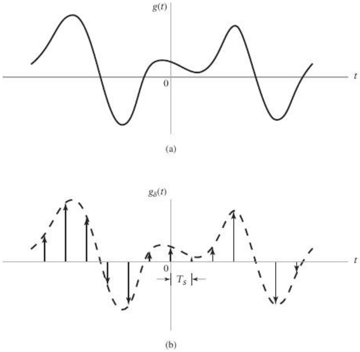
Today's class

- Sampling
- Pulse amplitude modulation
- Today's scribes are Kolli Aravind and Leen Roque Robin

Introduction

- Analog and digital pulse modulation
- Analog pulse modulation: A parameter of a periodic pulse train is varied
  - Amplitude, Width, or Position
- Digital pulse modulation: Coded pulses (e.g., presence or absence)
- In any form of pulse modulation, analog continuous time information is converted into a discrete time signal

Sampling



- Suppose $g(t)$ is a bandlimited finite energy signal
- The sequence $g[n] = g(nT_s)$ is obtained by "sampling" the signal at instants nT_s
- The sampling period is T_s and the sampling rate/frequency f_s is $\frac{1}{T_s}$
- This idealized model of sampling is called instantaneous sampling

Sampled signal - model

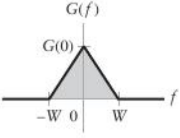
- We obtain a sequence $g[n] = g(nT_s)$ after sampling
- To analyze systems which are fed with sampled signals - either discrete time or continuous time analysis
- If continuous time analysis, then we use the following representation for the sampled signal
- $g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t - nT_s)$

Relationships between $g(t)$ and $g_\delta(t)$

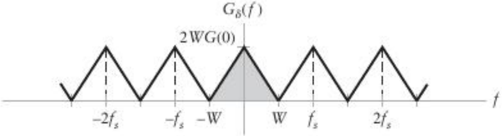
- What is the FT $G_\delta(f)$?

Relationships between $g(t)$ and $g_\delta(t)$

- What is the FT $G_\delta(f)$?
- $G_\delta(f) = f_s \sum_{m=-\infty}^{\infty} G(f - mf_s)$
- Suppose the signal $g(t)$ is bandlimited with bandwidth $2W$
- Suppose the sampling frequency is also $2W$.

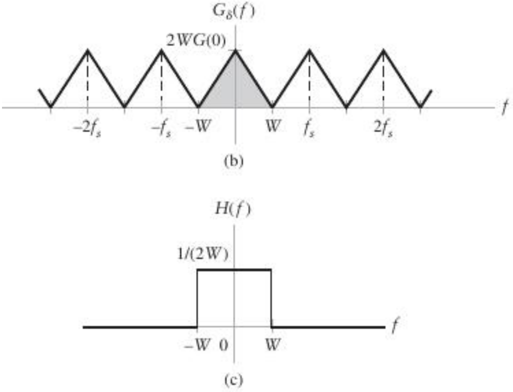


(a)



(b)

Relationships between $g(t)$ and $g_\delta(t)$



- In this case, it is possible to recover $g(t)$ from $g_\delta(t)$ using a reconstruction filter
- The reconstruction filter is non-causal. Read about the time-domain interpolation function from the text.

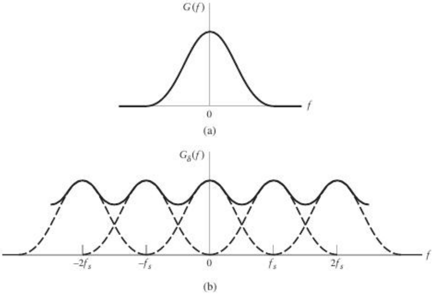
Sampling theorem

Theorem

A bandlimited signal $g(t)$ of finite energy with no energy in frequencies higher than W is completely specified by the samples $g(nT_s)$ where $T_s \leq \frac{1}{2W}$. Furthermore, $g(t)$ maybe recovered from its samples $g(nT_s)$ in this case.

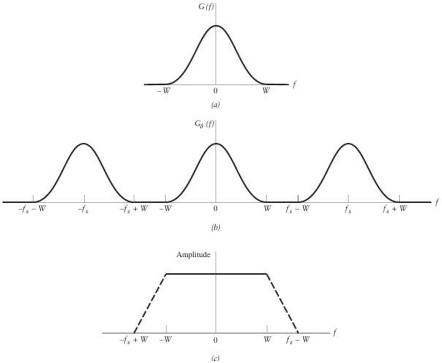
- $T_s \leq \frac{1}{2W}$ or $f_s \geq 2W$.
- The Nyquist sampling rate is $2W$

Aliasing



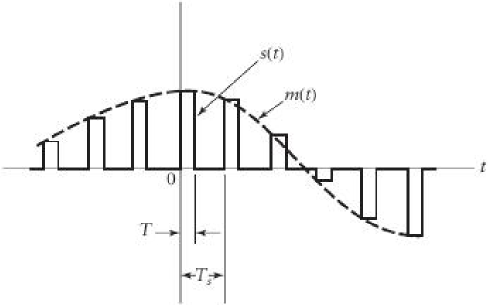
- Aliasing: Either $g(t)$ is not bandlimited or we "undersample" the signal $g(t)$

Aliasing



- Prefiltering using an anti-aliasing filter to bandlimit the signal that is actually sampled
- Sampling may be done at a $f_s > 2W$

Pulse amplitude modulation



- Message signal $m(t)$ is finite energy and bandlimited
- Sampling frequency is f_s which is greater than or equal to the Nyquist rate
- PAM signal $s(t) = \sum_{n=-\infty}^{\infty} m(nT_s)h(t - nT_s)$
- Note that this is different from $m(t) \times \sum_{n=-\infty}^{\infty} h(t - nT_s)$

Pulse amplitude modulation and demodulation system

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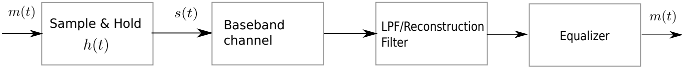

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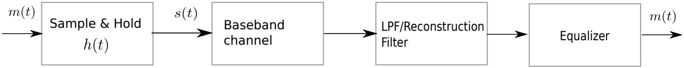
Pulse amplitude modulation and demodulation system



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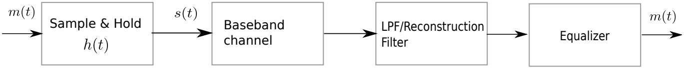
- So ... $m(t) \rightarrow$ Inst. sampling $\rightarrow m_\delta(t) \rightarrow h(t) \rightarrow s(t)$

Pulse amplitude modulation and demodulation system



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- So ... $m(t) \rightarrow$ Inst. sampling $\rightarrow m_\delta(t) \rightarrow h(t) \rightarrow s(t)$
- The equalizer has to compensate for $h(t)$
- The effect due to the $h(t)$ block is called Aperture effect