1 Signals and Spectra

Modulation

- Allows small antenna
- Can match with channel properties (e.g. optical fibre)
- Mixer/Filter/Amplifiers are typically better for high frequencies
- $0.01 < \frac{B}{f_c} < 0.1$ in a practical system
- A large bandwidth requires a large carrier frequency

Main Lobe BW of signal duration T_s

 T_s Duration × Bandwidth $\geq \frac{1}{4\pi}$ RMS

Real Signal

|M(-f)| = |M(f)| $\angle M(-f) = -\angle M(f)$

Fourier Series

 $s(t) = \sum_{n=0}^{\infty} c_n e^{j2\pi n f_0 t}$ $c_n = \frac{1}{T_0} \int_{-T_0/2}^{n=-\infty} s(t) e^{-j2\pi n f_0 t} dt \equiv c(nf_0) \quad E$ Fundamental Frequency $f_0 = \frac{1}{T_0}$

*n*th Harmonic $\pm nf_0$ For a real signal: $s(t) = c_0 + \sum_{\substack{n=1 \\ \forall t \in [0,5T_0, 0.5T_0]}}^{\infty} 2|c_n| \cos(2\pi n f_0 t + \angle c_n)$

$$S(f) = \int_{-\infty}^{\infty} s(t) e^{-j2\pi f t} dt$$

Inverse Fourier Transform $s(t) = \int_{-\infty}^{\infty} S(f) e^{j2\pi f t} df, \quad \forall t \in R$

 $\begin{array}{l} \textbf{3dB bandwidth} \\ 20 \log_{10} \Big(\frac{|S(f)|}{|S(f)|_{max}} \Big) \leq -3.01 (dB) \end{array}$

Sinusoid Power $\langle |\sin(t)|^2 \rangle = \langle |\cos(t)|^2 \rangle = \frac{1}{2}$

Average Power

$$1 \quad C^{\frac{T}{2}}$$

$$\langle s^2(t)\rangle = \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} s^2(t)dt, T \to \infty$$

Random Noise $S_N(f) = \begin{cases} 0.5N_0 & |f \pm f_c| \le B/2\\ 0 & \text{otherwise}\\ \left(\frac{S}{N}\right)_b = \frac{P_r}{N_0W} \end{cases}$

Channels 2

Distortionless Channel

 $y(t) = ks(t - t_0)$

Envelope/Group Delay $\tau_g := -\frac{1}{2\pi} \frac{d\angle H(f_0)}{df}$

Phase/Carrier Delay

$$\tau_p := -\frac{\angle H(f_c)}{2\pi f}$$

$$H_{eq}(f) = ke^{-j2\pi f t_0} H(f)^{-1}$$

Tapped Delay Filter $H_{eq}(f) = d_{M-n}e^{-j2\pi nf\Delta}$

$$\approx e^{-j2\pi f M\Delta} H(f)^{-1}$$

$$y(t) = \sum_{n=0}^{2M} d_{M-n} x(t - n\Delta)$$
$$\approx s(t - M\Delta)$$

$$\Delta < 0.5/W$$

3 Probability

Bayes' Rule $P(X|Y) \equiv \frac{P(X \cap Y)}{P(Y)}$ $\equiv \frac{P(Y|X)P(X)}{P(Y)}$

Expected Value
$$E[X] := \int_{-\infty}^{\infty} x f_x(x) dx$$

Moment order
$$n$$

 $[X^n] := \int_{-\infty}^{\infty} x^n f_x(x) dx$

Variance

$$\sigma_X^2 = E[(X - m_X)^2]$$
$$= E[X^2] - E[X]^2$$

Covariance

$$\operatorname{tov}[X, Y] := E[(X - m_x)(Y - m_y)]$$
$$= E[XY] - E[X]E[Y]$$

Correlation E[XY]

Correlation Coefficient

$$\sigma_{X,Y} := \frac{\operatorname{cov}[X,Y]}{\sigma_X \sigma_y} \in [-1,1]$$

Joint pmf

$$F_{X,Y}(x,y) := P(X \le x, Y \le y)$$

$$f_{X,Y}(x,y) := \frac{\delta^2}{\delta_x \delta_y} F_{X,Y}(x,y)$$

$$F_X(x) = F_{X,Y}(x,\infty)$$

$$f_x(x) = \int_{-\infty}^{\infty} f_{X,Y}(x,y) dy$$

$$F_{X,Y}(x,y) = \int_{-\infty}^{x} \int_{-\infty}^{y} f_{X,Y}(u,v) dvd$$
Conditional pdf
$$f_{Y|X}(y|x) = \frac{f_{X,Y}(x,y)}{f_X(x)}$$
Mutual Independence
$$f_{Y|X}(y|x) = f_Y(y)$$
Counsing PDE

Gaussian PDF

$$f_X(x) = \frac{1}{\sigma\sqrt{2\pi}} \exp(-\frac{(x-m)^2}{2\sigma^2}$$

$$> 0, x \in R$$

$$\Phi(x) := \text{cdf of N}(0,1)$$

$$Q(x) := 1 - \Phi(x)$$
,upper tail prob

Jointly Gaussian RVs Covariance matrix K = E[(X - m)(X - m)'] $f_x \equiv f_{X_1,\dots,X_n}(x_1,\dots,x_n)$

$$= \frac{\exp(-0.5(x-m)'K^{-1}(x-m))}{(2\pi)^{n/2}|\det K|^{1/2}}$$

Transformations of RVs If Y = g(X) is differentiable and $1\mathchar`-1$ then: $f_Y(y) = f_X(x) / \frac{dy}{dx}$ If Z = q(X, Y), W = h(X, Y)where the mapping

 $(g,h): \mathbb{R}^2 \to \mathbb{R}^2$ is 1-1 and differentiable then:

$$f_{Z,W}(z,w) = f_{X,Y}(x,y)/|\det J(x,y)$$
$$J(x,y) := \begin{bmatrix} \frac{\delta z}{\delta x} & \frac{\delta z}{\delta y} \\ \frac{\delta w}{\delta x} & \frac{\delta y}{\delta y} \end{bmatrix}$$

Central Limit Theorem

$$S_n := \sum_{\substack{i=1\\S_n - n\mu}} X_i$$
$$Z_N := \frac{S_n - n\mu}{\sigma\sqrt{n}}$$

$$\lim_{n \to \infty} P[Z_N \le z]$$
$$= \frac{1}{\sqrt{2\pi}} \int_{-\infty}^z \exp(-x^2/2) dx$$

Characteristic Function Essentially a Fourier transform

$$\Psi_X(v) := E[e^{jvX}]$$
$$= \int_{-\infty}^{\infty} e^{jvX} f_X(x) dx$$
If X,Y independent

$$\Psi_{X+Y}(v) = \Psi_X(v)\Psi_Y(v)$$
$$E[X^n] = \frac{1}{i^n} \frac{d^n \Psi_X(v)}{dv^n}|_{v=0}$$

E

3.1**Random Processes** Wide-sense Stationarity

• $m_X(t)$ constant for all time

• $R_X(s,t) \equiv R_X(s-t)$ acf depends only on time difference

Autocorrelation Properties Average Power $R_X(0) \equiv E[X(t)^2]$ Even $R_X(-\tau) = R_X(\tau)$ Max at $0 |R_X(\tau)| \le R_X(0)$ Power Spectral Density (PSD) $duS_X(f) := \int_{-\infty}^{\infty} R_X(\tau) e^{-j2\pi f\tau} d\tau$ Properties:

 $R_X(\tau) := \int_{-\infty}^{\infty} S_X(f) e^{j2\pi f\tau} df$ $E[X(t)^2] := \int_{-\infty}^{\infty} S_X(f) df$ (constant over t)

$$S_X(f) \ge 0$$

For real-valued process
 $S_X(f) = S_X(-f)$ White noise
 $S_X(f)$ is constant in f

Linear Systems with

Stationary Input
$$m_X \int^{\infty} h(\tau) d\tau = m_X$$

$$m_Y = m_X \int_{-\infty}^{\infty} h(\tau) d\tau = m_X H$$
$$R_Y(\tau) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} h(\tau_1) h(\tau_2) \cdot$$
$$R_X(\tau - \tau_1 + \tau_2) d\tau_1 d\tau_2$$
$$S_Y(f) = |H(f)|^2 S_X(f)$$

i.e Output PSD only depends on input PSD and system magnitude response

Cross-correlation

$$R_{XY}(s,t) := E[X(s)Y(t)] \equiv R_{YX}(t,s)$$

Cross-covariance

$$C_{XY}(s,t) := E[(X(s) - m_X(s)) \cdot (Y(t) - m_Y(t))]$$

$$\equiv C_{YX}(t,s)$$

Mutually uncorrelated:

$$X_{CY}(s,t) = 0$$

Cross Spectral Density

$$S_{XY}(f) := \mathbf{F}[R_{XY}(\tau)]$$

Bandpass Process

Real WSS random process is

- Lowpass: $S_X(f) = 0 \forall |f| >$ W, for some W > 0
- Bandpass: $S_X(f) = 0$ for all f outside $[-(f_0 + W), -(f_0 - W)]$ $W)]\&[f_0 - W, f_0 + W], \text{ for }$ some $f_0 > W \ge 0$

$$X(t) = X_c(t)\cos(2\pi f_0 t) - X_s(t)\sin(2\pi f_0 t)$$

 $X_c(t), X_s(t)$ lowpass, jointly WSS

$$X_c(t) = X(t)\cos(2\pi f_0 t) + \hat{X}(t)\sin(2\pi f_0 t)$$

$$X_s(t) = \hat{X}(t)\cos(2\pi f_0 t) - X(t)\sin(2\pi f_0 t)$$

 $R_{X_c}(\tau) = R_{X_s}(\tau)$ $= R_X(\tau)\cos(2\pi f_0\tau) + \hat{R}_X(\tau)\sin(2\pi f_0\tau)$

$$S_{X_c}(f) = S_{X_s}(f)$$

= $S_X(f - f_0) + S_X(f + f_0)$
for $|f| < W$

4 Trigonometry

$$\begin{aligned} \cos(x) &= \operatorname{Re}(e^{jx}) = \frac{e^{jx} + e^{-jx}}{2}\\ \sin(x) &= \operatorname{Re}(e^{jx}) = \frac{e^{jx} - e^{-jx}}{2j}\\ a\sin(\theta) + b\cos(\theta) &= R\sin(\theta + \alpha)\\ R &= \sqrt{a^2 + b^2} \quad \alpha = \arctan(\frac{b}{\alpha})\\ \sin(u \pm v) &= \sin u \cos v \pm \cos u \sin v\\ \cos(u \pm v) &= \cos u \cos v \mp \sin u \sin v\\ \sin(2u) &= 2\sin u \cos u\\ \cos(2u) &= \cos^2 u - \sin^2 u\\ &= 2\cos^2 u - 1\\ &= 1 - 2\sin^2 u\\ \sin u \sin v &= \frac{1}{2}[\cos(u - v) - \cos(u + v)]\\ \cos u \cos v &= \frac{1}{2}[\cos(u - v) + \cos(u + v)]\\ \sin u \cos v &= \frac{1}{2}[\sin(u + v) + \sin(u - v)]\\ \cos u \sin v &= \frac{1}{2}[\sin(u + v) - \sin(u - v)]\end{aligned}$$

5 Bandpass Signals and Systems

Any Bandpass signal can be alternatively written as

$$s(t) = a(t) \cos \underbrace{(2\pi f_c t + \varphi(t))}_{\theta(t)}$$
$$a(t) = \sqrt{s_c(t)^2 + s_s(t)^2}$$
$$\varphi(t) = \arctan(\frac{s_s(t)}{s_c(t)})$$

Bandpass Narrowband

 $f_0 >> B, \pm 0.5B$ of f_0

Bandpass to Lowpass (frequency)

1. Suprress the negative frequencies

 $Z(f) = (1 + \operatorname{sgn}(f))S(f)$

2. Downshift the spectrum

 $S_L(f) = z(f + f_0)$

Lowpass to Bandpass (frequency)

1. Upshift

$$Z(f) = S_L(f - f_0)$$

2. Reflect around 0, conjugate, add to PE and scale Z(t) + Z(-t)

 $S(f) = \frac{Z(f) + Z(-f)}{2}$

Bandpass to Lowpass (time)

$$z(t) \equiv s(t) + j\hat{s}(t)$$

2. Take IFT:

 $S_l(t) = e^{-j2\pi f_0 t} z(t)$ $z(t) = e^{j2\pi f_0 t} s_l(t)$

Lowpass to Bandpass (time)

 $s(t) = \operatorname{Re}[z(t)] = \operatorname{Re}[e^{j2\pi f_0 t} s_l(t)]$

Bandpass Representation

 $z(t) = A(t)e^{j(2\pi f_c t + \phi(t))}$ $s_l = s_c + js_s$

$$\operatorname{Re}(z(t)) = s(t) = A(t)\cos(2\pi f_c t + \phi(t))$$

$$= \underbrace{s_c(t)}_{\text{In phase}} \cos(2\pi f_0 t) - \underbrace{s_s(t)}_{\text{Quad}} \sin(2\pi f_0 t)$$

$$s_c(t) = s(t) \cos(2\pi f_0 t) + \hat{s}(t) \sin(2\pi f_0 t)$$

$$s_s(t) = -s(t) \sin(2\pi f_0 t) + \hat{s}(t) \cos(2\pi f_0 t)$$

Hilbert Transform

$$\begin{split} H(f) &= -j \operatorname{sgn}(f) \\ h(t) &= \frac{1}{\pi t} \\ \hat{s} &:= s * \frac{1}{\pi t} = \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{s(\tau)}{t - \tau} d\tau \\ \text{Common transform pairs} \\ \sin(t) \leftrightarrow -\cos(t) \end{split}$$

$$cos(t) \leftrightarrow sin(t)$$
6 Double Side Band

Suppressed Carrier (DSB-SC)

- Difficult to create an in phase local oscillator for demodulation
- Difficult to create an ideal mixer
 $$\begin{split} s(t) &= m(t)A_c\cos(2\pi f_c t) \\ S(f) &= 0.5A_c M(f) + 0.25A_c M(f-f_c) \\ &+ 0.25A_c M(f+f_c) \end{split}$$

Quadrature Carrier Multiplexing

Use a complex envelope to contain 2 real LP signals.

$$\begin{split} s(t) &= \mathrm{Re}\{z(t)\} = s_c(t) \cos(2\pi f_c t) - s_s(t) \sin(2\pi f_c t) \\ s_c(t) &= A_c m_1(t) \\ s_s(t) &= A_c m_2(t) \end{split}$$

Synchronous Demodulation

Multiply by $\cos(2\pi f_c t + \phi)$ where ϕ represents the phase offset due to an imperfect local oscillator.

 $v(t) = A_c m(t) \cos(2\pi f_c t)^2$

 $= 0.5A_c m(t)\cos(\phi)$

If $\phi \approx \pm \pi/2$, Quadrature Null Effect causes severe attenuation.

DSB-SC Power

 $P_{DSBSC} = 0.5A_c^2 P_m$

6.1 DSB-SC Noise Performance Received signal

 $S(t) = A_c M(t) \cos(2\pi f_c t) + N(t)$ $= (A_c M(t) + N_c(t)) \cos(2\pi f_c t)$

$$-N_{\rm s}(t)\sin(2\pi f_{\rm c}t)$$

$$= N_{s}(t) \sin(2\pi j c t)$$

$$S_{dem}(t) = 0.5A_C M(t) + 0.5N_c(t)$$
$$N'_c \text{ is LPFed in phase noise}$$
$$P_{dem} = 0.25A_c^2 E[M(t)^2] + 0.25E[N'_c(t)^2]$$
$$= 0.25A_c^2 E[M(t)^2] + 0.5N_0 W$$

Signal to noise ratio $\left(\frac{S}{N}\right)_{o} = \left(\frac{S}{N}\right)_{b} = \frac{A_{c}^{2}P_{M}}{2N_{0}W}$

7 Amplitude Modulation (AM)

• Uses more power than DSB-SC

Eliminates phase reversals

$$|m(t)| < \frac{1}{\mu}$$

 $s(t) = A_{e}(1 + \mu m(t)) \cos(2\pi f_{e}t)$

$$S(f) = 0.5A_c(\delta(f - f_c) + \mu M(f - f_c)) + 0.5A_c(\delta(f + f_c) + \mu M(f + f_c))$$

AM Demodulation

Use a notch filter to remove DC offset. Will be dangerous to perform this on signals with low frequency components due to the notch filter.

AM Power

 $P_{AM} = 0.5A_c^2 + 0.5\mu^2 A_c^2 P_m$

7.1 AM Noise Performance

Received signal

 $S(t) + A_c(1 + \mu M(t))\cos(2\pi f_c t) + N(t)$

Demodulated output (envelope detector)

$$\begin{split} Y(t) &= \sqrt{[A_c(1+\mu M(t))+N_c(t)]^2+N_s(t)^2]} \\ &\approx A_c(1+\mu M(t))+N_c(t) \end{split}$$

Notch filter to remove DC

Benjamin Ding

 $Y'(t) \approx \mu A_c M(t) + N_c(t)$

Signal to noise ratio

$$\frac{\left(\frac{S}{N}\right)_{o}}{\left(\frac{S}{N}\right)_{b}} = \frac{\mu^{2}P_{M}}{1+\mu^{2}P_{M}} < 1$$

Normalised Message
$$\left(\frac{S}{N}\right)_{o} = \frac{\mu^{2}P_{M_{N}}}{1+\mu^{2}P_{M_{N}}} \frac{P_{r}}{N_{o}W}$$

AM Threshold Effect

- If $N_0W >> A_c^2$ there is no meaningful SNR.
- Output SNR decreases linearly with power until A_c² is around N₀W at which signal quality suddenly decreases rapidly to zero

8 Single Side-Band (SSB)

- If the channel is LP with BW *B* Hz then there is no issue
- In a bandpass channel, after modulation the negative side-band uses valuable positive channel bandwidth
- Not suitable for low frequencies

Suppress negative sideband:

$$\tilde{M}(f) := (1 + \text{sgn}(f))M(f) = \begin{cases} 2M(f) &, f > 0\\ 0 &, f < 0 \end{cases}$$
$$\tilde{m}(t) = m(t) + j\left(\frac{1}{\pi t} * m(t)\right) = m(t) + j\hat{m}(t)$$
Spectrum:
$$S_{SSB}(f) = 0.5A_c\tilde{M}(f - f_c) + 0.5A_c\tilde{M}(-f - f_c)^*$$

$$= \begin{cases} A_c M(f - f_c) &, f > f_c \\ A_c M(f + f_c) &, f < -f_c \\ 0 & \text{elsewhere} \end{cases}$$

Hilbert Implementation(USB)

$$\begin{split} s_{SSB}(t) &= A_c m(t) \cos(2\pi f_c t) - A_c \hat{m}(t) \sin(2\pi f_c t) \\ &= \mathrm{Re}[\tilde{m}(t) A_c e^{j2\pi f_c t}] \end{split}$$

BPF Implementation (USB)

1. Generate DSB-SC Signal

$$S_{DSBSC}(f) = A_c M(f - f_c) + A_c M(f + f_c)$$

2. Pass through BPF with the passband $[f_c, f_c + B_{BPF}]$

Demodulation using Hilbert Transformer

Difficult to design a Hilbert transformer with sharp phase transition

$$s_c(t) = s(t)\cos(2\pi f_c t) + \hat{s}(t)\sin(2\pi f_c t) = A_c m(t)$$

Demodulation using LPF

Difficult to design a LPF with sharp magnitude transition.

1. Mix with carrier frequency

$$v(t) = s(t)\cos(2\pi f_c t)$$

$$= 0.5A_c m(t) + 0.5A_c m(t) \cos(4\pi f_c t) - 0.5A_c \hat{m}(t) \sin(4\pi f_c t)$$

SSB Signal at
$$2f_c$$
Hz

2. LPF with $W < B_{LPF} < 2f_c$

Using a local oscillator out of phase by a moderate ϕ is acceptable for telephony but not music/video with low frequency content

 $m_{rec}(t) = 0.5A_c(\cos(\phi)m(t) + \sin(\phi)\hat{m}(t))$ $M_{rec}(f) = 0.5A_c \begin{cases} e^{-j\phi}M(f) & f > 0\\ e^{j\phi}M(f) & f < 0 \end{cases}$

SSB Power

 $P_{SSB} = A_c^2 P_m$

Frequency Division Multiplexing

- Allows multiple messages on the same channel
- Put different bands of SSB modulation on the same carrier
- Crosstalk can be reduced with a guard band and using a low pass filter
- Use a bandpass filter for demodulation

SSB Noise Performance 8.1

Received signal

 $S(t) = A_c M(t) \cos(2\pi f_c t) - A_c \hat{M}(t) \sin(2\pi f_c t) + N(t)$

Demodulated output

 $Y(t) = 0.5A_c M(t) + 0.5N_c(t)$

Noise power

 $P_{N_C} = N_0 W$

Signal to noise ratio

 $\left(\frac{S}{N}\right)_{o} = \frac{A_{c}^{2} P_{M}}{N_{0} W} = \frac{P_{R}}{N_{0} W}$

9 Vestigial Side Band (VSB)

- Compromise between SSB and DSB-SC
- Partially suppress most of lower (or upper) sidebangs except a vestige
- For demodulation via envelope detection β cannot be too small

 $H_{vsb}(f) = \begin{cases} 2 & f_c + \beta \le f \le f_c + W \\ 0 & 0 \le f \le f_c - \beta \end{cases}$

 $H_{vsb}(f_c + w) + H_{vsb}(f_c - w) = 2$

Cases:

SSB: $2\beta \approx 0$

DSB-SC: $\beta >> W$, with two sidebands and no quadrature component

Demodulation using local oscillator/LPF

1. Mix with local oscillator

2. BPF $(f_m > W)$ or will overlap!

$$S_{dem}(f) \propto [S_{VSB}(f+f_c) + S_{VSB}(f-f_c)]H_{LPF}(f)$$
$$\propto [H_{VSB}(f+f_c) + H_{VSB}(f-f_c)]M(f)$$
$$= 2M(f)$$

Demodulation using envelope detection

Can be combined with AM: VSB + C

$$s(t) = A_c(1 + \mu m(t))\cos(2\pi f_c t) - \underbrace{A_c \mu \gamma(t)}_{c} \sin(2\pi f_c t)$$

$$\therefore \text{Envelope} \propto A_c \sqrt{(1+0.5\mu m(t))^2 + (\mu\gamma(t))^2}$$

For a small μ :
Envelope $\propto 1 + \mu m(t)$

Transition Bandwidth Selection

- When transition bandwidth $2\beta \approx 0$, VSB resembles SSB
- When $\beta >> W$, VSB filter looks like ideal BPF centered at the carrier frequency and hence the signal is approximately a DSB-SC signal.
- For envelope detection β should not too small

10 Angle Modulation

Only the angle contains message information

 $s(t) = A_c \cos(\theta(t)) = A_c \cos(2\pi f_c t + \phi(t))$

Phase Modulation (PM) 10.1

Phase of carrier is modulated in proportion to message

$$\phi(t) = k_p m(t)$$

 $s_{PM}(t) = A_c \cos(2\pi f_c t + k_p m(t))$

 k_p : Phase Sensitivity Phase modulation index/Maximum Phase Deviation $\beta := \max_{t} |\phi(t)| = k_p \max |m(t)|$

11 **Frequency Modulation** (FM)

$$\begin{split} s_{FM}(t) &= A_c \cos(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau) \\ m(\tau) &= 0, \forall \tau < 0 \\ \phi(t) &= 2\pi k_f \int_0^\tau m(\tau) d\tau \end{split}$$
Instantaneous Frequency: $f_i(t) = \frac{1}{2\pi} \frac{d\theta}{dt} = f_c + k_f m(t)$ Maximum Frequency Deviation $\Delta f := k_f \max[m(t)]$

Single Tone Analysis

 $m(t) = A_m \cos(2\pi W t)$

Frequency Modulation Index $\beta = \frac{\Delta f}{W} = \frac{k_f \max(m(t))}{W}$

 S_F

$$M(t) = \operatorname{Re}\{A_c e^{j2\pi k_f} \int_0^t m(\tau) d\tau + j2\pi f_c t\}$$
$$= A_c \cos(\beta \sin(2\pi W t)) \cos(2\pi f_c t)$$

 $-A_c \sin(\beta \sin(2\pi W t)) \sin(2\pi f_c t)$

Narrow-Band FM

In the case where:

$$\beta = \frac{\Delta f}{W} = \frac{k_f A_m}{W} << 1$$

$$\cos x \approx 1 \quad \sin x \approx 1$$

The magnitude spectrum looks like AM, but unlike AM the quadrature carrier is modulated

 $s_{FM}(t) \approx A_c \cos(2\pi f_c t) - 0.5 A_c \beta \cos(2\pi (f_c - W)t)$ $+0.5A_c\beta\cos(2\pi(f_c+W)t)$

cannot make assumption

$$s_{FM}(t) = \operatorname{Re}\{A_c e^{j2\pi f_c t + \phi(t))}\}$$
$$= A_c \sum_{n = -\infty}^{\infty} J_n(\beta) \cos(2\pi (f_c + nW)t)$$

Bessel Function If $\beta << 1$

 $J_0(\beta) \approx 1, J_1(\beta) \approx \beta, J_n(\beta) \ll \beta$

FM Power

Independent of message

 $P_{FM} = 0.5A_c^2$

Carson's Rule

Estimation of the required FM bandwidth, works well for $\beta >> 1$ and $\beta << 1$. Underestimates in range closer to β $B_{Carson} = 2(1+\beta)W = 2(W+\Delta f)$

Direct FM Generation

- At high frequency, use a voltage-controller oscillator (VCO).
- At low frequency, use a parallel resonant circuit by modulating the capacitance of the circuit. At the resonant frequency

Indirect FM Generation

Major disadvantage is drifting carrier frequency

- 1. Generate narrow-band FM with stable freq using standard linear modulation techniques
- 2. Frequency multiplication to generate wideband FM using a non-linear device

Non-linearities in FM

- Strong: when non-linearity is inserted intentionally
- Weak: when non-linearity arises whenever signal levels become too large. FM is impervious to weak non-linearities

11.1 **Frequency Demodulation**

Deviation ratio

Strength of the demodulated signal increases with deviation ratio.

 $D = \Delta f / W (= \beta \text{ if single tone})$

FM Power

$$P_{FM} = 0.5A$$

11.1.1 FM-AM Demodulation

Using differentiator

$$\frac{ds_{FM}(t)}{dt} = A_c (2\pi f_c + 2\pi k_f m(t)) \cdot \\ \sin(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau) \\ F\left\{\frac{ds_{FM}(t)}{dt}\right\} = j2\pi f S_{FM}(f)$$

Slope Circuit

Wide Band Single Tone FM The differentiator amplitude response at the Unlike in narrow band, β is not small and carrier frequency is too large so a slope circuit

2

$$\sin(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau)$$
$$F\left\{\frac{ds_{FM}(t)}{dt}\right\} = j2\pi f S_{FM}(f)$$

Benjamin Ding

is needed.

$$H(f) = \begin{cases} j2\pi a(f - f_c + C) & |f - f_c| < B/2 \\ H(-f)^* & |f + f_c| < B/2 \\ 0 & \text{otherwise} \end{cases}$$

Output of demodulator:

$$y(t) = -2\pi a A_c (C + k_f m(t)) \sin(2\pi [f_c t + k_f \int_0^t m(\tau) d\tau]) R(t)$$

- $C \ge k_f max_t |m(t)| = \Delta f$ for enveloped detection, common choice is C = B/2
- Demodulated signal amplitude $\propto k_f$

Limiter

- Limits FM signal with constant amplitude
- Used to avoid distorted message due to amplitude distortion in slope circuit of PLL demodulation.

 $v_o(t) = A' \operatorname{sgn}(v_i(t))$

FM signal with constant amplitude

11.1.2 Zero Crossings Detector Demodulation

 $f_c + k_f m(t) \approx \frac{\text{no. zero crossings in interval}}{2T}$

- 1. Limit FM signal (easier to detect zeros)
- 2. Apply pulse-detector pulses for each positive/negative transition
- 3. Continuously integrate over interval T. The integral is proportional to the number of pulses in T.

11.1.3 Feedback Frequency Demodulation

Phase locked loop

VCO with frequency sensitivity k_v and free running frequency f_v

$$t) = -\sin\left(\frac{2\pi f_v t + 2\pi k_v \int_0^t y(\tau) d\tau}{=:\theta_v(t)}\right)$$

Multiply received FM signal with VCO output

 $x(t) = -A_c \cos(\theta_c(t)) \sin(\theta_v(t))$

v(

$$= 0.5A_c \left(\underbrace{\sin(\theta_c(t) - \theta_v(t))}_{\text{low difference freq}} - \underbrace{\sin(\theta_c(t) + \theta_v(t))}_{\text{high sum freq}} \right)$$

- 1. Pass through LPF to suppress the high frequency term.
- 2. Amplify phase detector output by a factor K_a
- 3. Feed back into VCO to keep angle difference between VCO and FM small $\epsilon := \theta_c(t) - \theta_v(t)$

Well designed PLL keeps angle difference small so can approximate linearly

$$\begin{split} \dot{z}(t) &= 2\pi (f_c - f_v + k_f m(t) - k_v 0.5 K_a A_c \sin(\epsilon(t))) \\ \therefore y(t) &\approx 0.5 K_a A_c \epsilon(t) \approx \frac{f_c - f_v + k_f m(t)}{k_v} \\ & \text{Ensure } f_c - f_v, \Delta f << 0.5 k_v K_a A_c \end{split}$$

11.2 FM Noise Performance

• Output SNR increases with D at cost of increased transmission BW B = 2(D+1)W

- Output SNR increases with A_c quadratically, increasing carrier amplitude decreases output noise power
- Dynamic range compression on message increases $\frac{P_M}{\max_t |m(t)|^2}$ improving output SNR

Received signal

$$[a\tau]) R(t) = A_c \cos(2\pi f_c(t) + \Phi(t)) + N(t)$$

Noise decomposition

$$N(t) = N_c(t)\cos(2\pi f_c t) - N_s(t)\sin(2\pi f_c t)$$
$$\equiv V_n(t)\cos(2\pi f_c t + \phi_n(t))$$

After Limiter and BPF

$$\begin{split} S(t) &= A\cos(2\pi f_c t + \phi(t) + \angle G(t)) \\ G(t) &:= 1 + \frac{V_n(t)}{A_c} e^{j(\phi_n(t) - \phi(t))} \end{split}$$

Demodulator output

$$Y(t) = 2\pi k_f M(t) + \frac{d\angle G(t)}{dt}$$

has a nonlinear dependence on messa

 $\angle G(t)$ has a nonlinear dependence on message and is difficult to analyse

> Large Carrier Approximation Assume

$$\begin{aligned} E[V_n(t)^2] &= 2N_0B << A_c^2\\ G(t) &\approx \frac{N_s(t)\cos(\phi(t)) - N_c(t)\sin(\phi(t))}{A_c} \end{aligned}$$

L

Wideband Approximation

If $W \ll B/2$ then for all $|\tau| < 1/W$ $R_{\angle G}(\tau) \approx R_{N_C}(\tau)/A_c^2$

$$\begin{array}{l} \textbf{Output Noise PSD}\\ S_Z(f) = \begin{cases} (2\pi f)^2 \frac{N_0}{A_c^2} & |f| < W\\ 0 & |f| > W\\ P_Z = \frac{8\pi^2 N_0 W^3}{3A_c^2} \end{array}$$

Signal to noise ratio

$$\begin{split} \left(\frac{S}{N}\right)_o &= \frac{3A_c^2k_f^2P_M}{2N_0W^3} \\ &= \frac{3D^2P_M}{\max_t |m(t)|^2} \left(\frac{S}{N}\right)_b \end{split}$$

Maximum Deviation Ratio

 $20(D+1) < 10^{(S/N)_b/10}$

Clicks

- If $N_0B \ll 0.5A_c^2$ is not satisfied, significant probability that noise envelope $V_n(t) \ge A_c$.
- Small variations in-phase and quadrature noise will occasionally lead to large phase changes of $\pm 2\pi$
- Demodulator differentiates large phase changes yielding impulse like effects heard as clicks

Threshold Effect

To avoid falling below SNR threshold

$$\begin{split} & \left(\frac{S}{N}\right)_b > 20(1+D) =: \left(\frac{S}{N}\right)_{b,th} \\ & \left(\frac{S}{N}\right)_o > \frac{60P_M}{\max_t |m(t)|^2} D^2(1+D) =: \left(\frac{S}{N}\right)_{o,th} \end{split}$$

$$\begin{split} \left(\frac{S}{N}\right)_{o} &= \frac{\mathbf{FM} \; \mathbf{Parameter} \; \mathbf{Check}}{\max_{t} \; |m(t)|^{2}} \frac{P_{r}}{N_{0}W} \geq \left(\frac{S}{N}\right)_{o,desired} \\ & 20(D+1) \leq \frac{P_{r}}{N_{0}W} \\ & 2W(D+1) \leq B_{avail} \end{split}$$

Pre-emphasis/De-emphasis

- High frequencies won't sound as clean due to quadratic FM output noise PSD
- Put message through highpass filter before modulation to emphasis high frequency components
- Pass demodulated signal through lowpass filter to attenuate noise at high frequencies

$$\begin{aligned} H_d(f) &\equiv \frac{1}{1+jf/f_0} \\ H_e(f) &= 1/H_d(f) = 1+jf/f_0 \\ \text{New output noise PSD} \\ S_{Z,pd}(f) &= 4\pi^2 \frac{N_0}{A_c^2} \frac{f^2}{1+(f/f_0^2)} \\ P_{Z,pd} &= 8\pi^2 \frac{N_0 f_0^3}{A_c^2} \left(\frac{W}{f_0} - \arctan\left(\frac{W}{f_0}\right)\right) \\ \text{Ratio of new SNR to old} \\ \left(\frac{S}{N}\right)_{o,pd} / \left(\frac{S}{N}\right)_o &= \frac{1}{3} \left(\frac{(W/f_0)^3}{W/f_0 - \arctan(W/f_0)}\right) \end{aligned}$$

12 Digital Communications

Pulse Amplitude Modulation

$$\mathbf{s}(t) = \sum_{n=-\infty}^{\infty} d_n p(t - nT)$$

 d_n Amplitude of pulse

T Sampling period

p(t) Signalling waveform (pulse) Ideal Sampler

Outputs a modulated sequence of impulses

$$x_{delta}(t) := \sum_{n=-\infty}^{\infty} x(nT_s)\delta(t - nT_s)$$
$$X_{delta}(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - \frac{n}{T_s})$$

Impulse Train Fourier Series

$$\dot{a}(t) = \sum_{m=-\infty}^{\infty} \delta(t - mT_s) = \sum_{n=-\infty}^{\infty} c_n e^{j2\pi nt/T_s}$$
$$c_n \equiv \frac{1}{T_s}$$

12.1 Quantisation

Quantisation Quality

Measured by mean square error (MSE)

$$D = E[(X - Q(X))^2]$$

Signal to quantisation noise ratio

$$SQNR := \frac{E[X^2]}{E[(X - Q(X))^2]}$$

Uniform Quantisation

If X U[a, b), minimise D by partitioning [a, b)into N intervals of equal length

$$\begin{split} N &= \frac{b^2}{b-a} \\ \Delta &:= \frac{b^2}{a} \\ \hat{x} &= \text{Mid-point} \\ E[(X - Q(X))^2] &= \frac{\Delta^2}{12} = \frac{E[X^2]}{N^2} \\ \text{Optimal SQNR} &= N^2 \end{split}$$

For nonuniform X, for
$$N >> 1$$
 still have
 $\Lambda^2 = F[X^2]$

 $E[(X - Q(X))^2] \approx \frac{\Delta^2}{12} = \frac{E[X^2]}{N^2}$ given $P[X < a \cup X > b]$ is comparatively small

Companding Quantisation

Using a uniform quantiser to generate

$$\hat{x} = g^{-1}(Q_U(g(x)))$$

Where g(x) is a compressor, $g^{-1}(x)$ is an expander.

 $\mu extsf{-Law}$

$$g(x) = \frac{\ln(1+\mu|x|)}{\ln(1+\mu)}\operatorname{sgn}(x)$$

A-Law

$$g(x) = \begin{cases} \frac{Ax}{1+\ln A} & |x| < 1/A\\ \frac{1+\ln(A|x|)}{1+\ln A} \operatorname{sgn}(x) & |x| \ge 1/A \end{cases}$$

12.2 Waveform Coding

Pulse Code Modulation

- 1. Pass cont. time signal through anti aliasing LPF
- 2. Sample and hold at sampling rate $\geq 2W$
- 3. Quantise each sample
- 4. Encode each $N \equiv 2^v$ level as v-bit word
- 5. Convert to serial bit stream, bit rate $r=1/T_b$
- 6. Generate regular signalling pulses once every Ts and use bit stream to modulate their amplitudes

Differential PCM

- For a fixed number of quantisation levels N, quantisation noise power increases with variance of signal
- Oversample signal at $f_s > 2W$, successive samples highly correlated and have little variance
- Store last quantised estimate $\hat{X}((k-1)T_s)$ in a delay buffer and quantise $X(kT_s) - \hat{X}((k-1)T_s)$ for higher resolution

At Transmitter

- 1. At time k find $\hat{Y}_k = Q(X_k \hat{X}_{k-1})$ where $\hat{X}_{-1} := m_X$
- 2. Encode \hat{Y}_k into bits and transmit using PCM
- 3. Update signal estimate $\hat{X}_k = \hat{X}_{k-1} + \hat{Y}_k$ At Receiver

At Receiver

- 1. Demodulate received PCM signal to get \hat{Y}_k
- 2. Update signal estimate via $\hat{X}_k = \hat{X}_{k-1} + \hat{Y}_k$ where $\hat{X}_{-1} := m_x$

Delta Modulation

- Extreme case of DPCM where a 1-bit quantiser is used $\hat{Y}_k = \pm \Delta$
- Useful when can sample much faster than Nyquist
- Multi level A/D conversion can be slow so this method can be effective instead

- If Δ too large, slow input changes will oscillate
- If Δ too small, delta modulator can't keep pace with fast input changes and gives sustained under/over shoot
- Adaptive ΔM : increase Δ if the last few \hat{Y}_k have had the same sign. E.g.

$$\Delta_k = \Delta_{k-1} (1.5)^{\operatorname{sgn}(Y_k Y_{k-1})}$$

12.3 Detecting Binary PCM

- Above PCM threshold, extra transmit power does not improve SNR
- SNR gain to bandwidth increase is exponential relationship
- Analog modulation is a better option than high v PCM if enough baseband SNR is available
- Can use error control coding to achieve optimal performance

Assume channel bandwidth sufficient for pulse

and only adds uncorrelated white noise

$$R(t) = dp(t) + N(t), 0 \le t \le T_b$$

$$R(t) = dp(t) + N(t), 0 \le t < T_0$$

Noise Limiting Filter

• Pass received signal through non-ideal filter H(f) to limit noise power

 $y(t) = d(h * p)(t) + (h * N)(t), 0 \le t < T_b$

 Distortion of pulse from the filter is fine, as long as at the *decision instant* t' > 0, |(h * p)(t')| is large compared to filtered noise (h * N)(t')

Optimal Detection Filter

Desired condition for low probability of misdetection

 $|d(h * p)(t')|^2 << \operatorname{var}[(h * N)(t')], d \neq 0$

Find h(t) to maximise ratio at decision instant

 $\frac{d^2 |(h*p)(t')|^2}{E[|h*N)(t')|^2]}$

Matched Filter

Find
$$H(f)$$
 to maximise

$$\eta = \frac{|\int_{-\infty}^{\infty} H(f)P(f)e^{j2\pi ft'}df|^2}{\int_{-\infty}^{\infty} |H(f)|^2 df}$$
$$\eta_{max} = \int_{-\infty}^{\infty} |P(f)|^2 df$$
$$H_{opt}(f) = e^{-j2\pi ft'}P(f)^*$$
$$h_{opt}(t) = p(t'-t)$$
Noise PSD at output

$$E[|(h * N)(t')|^2] = 0.5N_0 \int_{-\infty}^{\infty} |H(f)|^2 df$$

Optimal Output Detection SNR

It can be seen the energy of
$$p(t)$$
 is important,
not its peak value

$$\left(\frac{S}{N}\right) = \frac{2d^2 E_p}{N_0}$$

Statistical Properties of Matched Filter Output Mean C^{T_b}

$$Mean \\ E[Y(T_b)|d] = \int_0^{T_b} dp(t)dt = dE_p$$

Variance
$$ar[Y(T_b)|d] = 0.5N_0E_p$$

Probability of Error
If '0' transmitted

$$p_{e0} = Q\left(\frac{V - d_0 E_p}{\sigma_z}\right)$$

If '1' transmitted
 $p_{e1} = 1 - Q\left(\frac{V - d_1 E_p}{\sigma_z}\right)$

Average probability of error
$$= p_0 Q \left(\frac{V - d_0 E_p}{\sigma_z} \right) + p_1 \left(1 - Q \left(\frac{V - d_1 E_p}{\sigma_z} \right) \right)$$

Optimal threshold

 p_{ϵ}

For equally likely bits $p_0 = p_1 = 0.5$ $v_{opt} = \frac{(d_0 + d_1)E_p}{2}$ $p_e = Q\Big((d_1 - d_0)\sqrt{\frac{E_p}{2N_0}}\Big)$

Error Probability vs Average Power

Average energy per bit

$$P_r T_b := E_b$$
Unipolar: $P_r = 0.5A^2 E_p/T_b$

$$p_e = Q\left(\sqrt{\frac{P_r T_b}{N_0}}\right)$$
Polar: $P_r = 0.25A^2 E_p/T_b$

$$p_e = Q\left(\sqrt{\frac{2P_r T_b}{N_0}}\right)$$

Polar signalling has better performance for given power

Natural Binary Coding

If a natural binary format is used and

$$(S/N)_b >> v(>> 1)$$

Output noise power

 $\approx \underbrace{\frac{|x|_{max}^2}{4^v 3}}_{\text{From Quantisation Noise}} + \underbrace{\frac{4|x|_{max}^2 p_e}{3}}_{\text{From Channel Noise}} \\ \left(\frac{S}{N}\right)_o \approx \frac{3(P_X/|x|_{max}^2)}{4^{-v} + 4Q\left(\sqrt{\frac{(S/N)_b}{v}}\right)}$

If $(S/N)_b >> v^2$ quantisation noise dominates

13 Inter-Symbol Interference

Given a distorting channel H(f) which also adds Gaussian white noise

$$\begin{split} \tilde{p}(t) &:= (p * h * m)(t) \\ Z(t) &:= (h * m * N)(t) \\ \text{Filter output given by} \\ t) &= \sum_{n=0}^{\infty} d_n \tilde{p}(t - nT_b) + Z(t) \end{split}$$

If we want to recover
$$d_{k-1}$$
 transmitted during
previous bit period, the output of the filter is

$$Y(kT_b) = \underbrace{d_{k-1}\tilde{p}(T_b)}_{\text{Desired}} + \underbrace{\sum_{\substack{n:n \neq k-1 \\ \neq 0 \to \text{ISI}}} d_n \tilde{p}((k-n)T_b)}_{\text{Poise}} + \underbrace{Z(kT_b)}_{\text{Noise}}$$

Can mitigate using Pulse Shaping or Equalisation

$f(t) = \mathcal{F}^{-1} \{ F(f) \} = \int_{-\infty}^{+\infty} f(t) e^{j2\pi f t} df$	$\overset{\mathcal{F}}{\longleftrightarrow}$	$F(f) = \mathcal{F}\left\{f(t)\right\} = \int_{-\infty}^{+\infty} f(t)e^{-j2\pi ft}dt$
transform $f(t)$ time reversal $f(-t)$ complex conjugation $f^*(t)$ reversed conjugation $f^*(-t)$	$\begin{array}{c} \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \end{array}$	$F(f)$ frequency reversal $F(-f)$ reversed conjugation $F^*(f)$ complex conjugation
$\begin{array}{c} f(t) \text{ is purely real} \\ f(t) \text{ is purely imaginary} \\ \text{even/symmetry} & f(t) = f^*(-t) \\ \text{odd/antisymmetry} & f(t) = -f^*(-t) \end{array}$	$\begin{array}{c} \xrightarrow{\mathcal{F}} \\ \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\$	$F(f) = F^*(-f)$ even/symmetry $F(f) = -F^*(-f)$ odd/antisymmetry $F(f)$ is purely real $F(f)$ is purely imaginary
time shifting $f(t-t_0)$ $f(t)e^{j2\pi f_0 t}$ time scaling $f(af)$ $\frac{1}{ a }f\left(\frac{f}{a}\right)$	$\begin{array}{c} \mathcal{F} \\ \mathcal{F} \end{array}$	$F(f)e^{-j2\pi f t_0}$ $F(f - f_0) \qquad \text{frequency shifting}$ $\frac{1}{ a }F\left(\frac{f}{a}\right)$ $F(af) \qquad \text{frequency scaling}$
linearity $af(t) + bg(t)$ time multiplication $f(t)g(t)$ frequency convolution $f(t) * g(t)$	$\begin{array}{c} \xrightarrow{\mathcal{F}} \\ \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \end{array}$	aF(f) + bG(t) F(f) * G(f) frequency convolution F(f)G(f) frequency multiplication
delta function $\delta(t)$ shifted delta function $\delta(t-t_0)$ 1 $e^{j2\pi f_0 t}$	$\begin{array}{c} \xrightarrow{\mathcal{F}} \\ \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\$	$\begin{array}{l} 1 \\ e^{-j2\pi ft_0} \\ \delta(f) \\ \delta(f-f_0) \end{array} \qquad \qquad \text{delta function} \end{array}$
two-sided exponential decay $e^{-a t }$ $a > 0$ $e^{-\pi t^2}$ $e^{j\pi t^2}$	$\begin{array}{c} \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \end{array} \end{array}$	$\frac{2a}{a^2+4\pi^2 f^2} \\ e^{-\pi f^2} \\ e^{j\pi(\frac{1}{4}-f^2)}$
sine $sin (2\pi f_0 t + \phi)$ cosine $cos (2\pi f_0 t + \phi)$ sine modulation $f(t) sin (2\pi f_0 t)$ cosine modulation $f(t) cos (2\pi f_0 t)$ squared sine $sin^2 (t)$ squared cosine $cos^2 (t)$	$\begin{array}{c} \xrightarrow{\mathcal{F}} \\ \xleftarrow{\mathcal{F}} \\$	$ \frac{j}{2} \left[e^{-j\phi} \delta(f+f_0) - e^{j\phi} \delta(f-f_0) \right] \frac{1}{2} \left[e^{-j\phi} \delta(f+f_0) + e^{j\phi} \delta(f-f_0) \right] \frac{j}{2} \left[F(f+f_0) - F(f-f_0) \right] \frac{1}{2} \left[F(f+f_0) + F(f-f_0) \right] \frac{1}{4} \left[2\delta(f) - \delta\left(f - \frac{1}{\pi}\right) - \delta\left(f + \frac{1}{\pi}\right) \right] \frac{1}{4} \left[2\delta(f) + \delta\left(f - \frac{1}{\pi}\right) + \delta\left(f + \frac{1}{\pi}\right) \right] $
rectangularrect $\left(\frac{t}{T}\right) = \begin{cases} 1 & t \leqslant \frac{T}{2} \\ 0 & t > \frac{T}{2} \end{cases}$ triangulartriang $\left(\frac{t}{T}\right) = \begin{cases} 1 - \frac{ t }{T} & t \leqslant T \\ 0 & t > T \end{cases}$ step $u(t) = 1_{[0,+\infty]}(t) = \begin{cases} 1 & t \ge 0 \\ 0 & t < 0 \end{cases}$ signum $\operatorname{sgn}(t) = \begin{cases} 1 & t \ge 0 \\ -1 & t < 0 \end{cases}$	$\begin{array}{c} \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \end{array}$	$T \operatorname{sinc} Tf$ $T \operatorname{sinc}^2 Tf$ $\frac{1}{j2\pi f} + \delta(f)$ $\frac{1}{j\pi f}$
sincsinc (Bt) squared sincsinc ² (Bt)	$ \stackrel{\mathcal{F}}{\longleftrightarrow} $	$\frac{\frac{1}{B}\operatorname{rect}\left(\frac{f}{B}\right) = \frac{1}{B}1_{\left[-\frac{B}{2}, +\frac{B}{2}\right]}(f)$ $\frac{1}{B}\operatorname{triang}\left(\frac{f}{B}\right)$
n-th time derivative $\frac{d^n}{dt^n}f(t)$ n-th frequency derivative $t^n f(t)$ $\frac{1}{1+t^2}$	$\begin{array}{c} \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \\ \stackrel{\mathcal{F}}{\longleftrightarrow} \end{array}$	$(j2\pi f)^n F(f)$ $\frac{1}{(-j2\pi)^n} \frac{d^n}{df^n} F(f)$ $\pi e^{-2\pi f }$

Table of Continuous-time Frequency Fourier Transform Pairs