Two types of CDMA systems deployed in practice:

→ direct sequence spread spectrum (DSSS)

→ frequency hopping spread spectrum (FHSS)

Direct sequence spread spectrum (DSSS):

→ what we studied using linear algebra

Each user gets their own vector

→ called code vector (i.e., private key)

→ code vectors between users: orthogonal

→ in practice: also look random (pseudo-random)

→ prevents easy eavesdropping

Additional features/variations of DSSS:

Replication: replicate each data bit r-fold

→ ex.: if $r = 3$ and data is 1001, then 111000000111

→ why?

Scramble data bits: one-time pad idea

Ex.:

- data bits 1000100000
- pseudo-random bits 1010011010

  → private key or one-time pad

  → called chipping sequence
- compute: data bits XOR chipping sequence

  → $1000100000 \oplus 1010011010 = 0010111010$

  → achieves one-time pad encryption to prevent eavesdropping

- sender transmits XOR'ed bit sequence 0010111010

  → e.g., use amplitude (and/or phase) modulation of carrier wave

  → separate issue

- how does receiver decode sender's data bits?

Other benefit of scrambling with pseudo-random key?

→ hint: why it's called "spread spectrum"

Single-user DSSS: used in 802.11b WLAN

→ 11-bit chip sequence

→ single-user means: two laptops do not use orthogonal code vectors for simultaneous bit transmission

→ even use same chip sequence

→ multi-user communication: a different method called CSMA (carrier sense multiple access)

→ similar to Ethernet's method

→ discussed under link layer protocols

Second type of CDMA: frequency hopping spread spectrum (FHSS)

Select m carrier frequencies $f_1, f_2, \ldots, f_m$.

→ e.g., $m = 5$ with $f_1 = 101$ MHz, $f_2 = 102$ MHz, …, $f_5 = 105$ MHz

To send k bits, select pseudo-random sequence from 1, 2, …, m of length k

→ e.g., if $k = 10$ then 3 5 2 1 4 2 5 3 4 1

Send first bit on frequency f_3, second bit on f_5, …, 10th bit f_1.

→ hop around

→ pseudo-random sequence is like private key

Benefits:

→ prevents eavesdropping

→ resistant to jamming ("spread spectrum")

Drawback?

Used in old IEEE 802.11 WLAN (standards specify both DSSS and FHSS)

Used in old IEEE 802.11 Bluetooth

→ 79 frequency hopping sequence

→ now: part of 802.15

→ wireless PAN (personal area network)

Back to orthogonal FDM (OFDM)

→ key idea: use carrier waves that are orthogonal

→ spectra of carrier frequencies can overlap without causing interference

→ old guard band spacing not necessary

What does it mean for sine waves to be orthogonal to each other?

Dot product of two vectors $x = (x_1, \ldots, x_n)$ and $y = (y_1, \ldots, y_n)$

$$x \circ y = \sum_{i=1}^{n} x_i y_i$$

$\longrightarrow$ sum of products

Dot product of two sinusoids $x(t) = \sin f_x t$ and $y(t) = \sin f_y t$

$$x(t) \circ y(t) = \int_{-\infty}^{\infty} (\sin f_x t)\,(\sin f_y t)\,dt$$

$\rightarrow$ again: just a sum of products

$\rightarrow$ ex.: $\sin t$ and $\sin 2t$

More generally: $x(t) \circ y(t) = \int_{-\infty}^{\infty} e^{if_x t} e^{-if_y t} dt$

$\rightarrow$ since Fourier transform involves complex sinusoids

$\rightarrow$ but same form: sum of products

How to get N mutually orthogonal sinusoids?

- Suppose available frequency lies between f_a and f_b

  $\rightarrow$ bandwidth: $W = f_b - f_a$

  $\rightarrow$ ex.: $f_a = 2.4$ GHz, $f_b = 2.5$ GHz, $W = 100$ MHz

- Choose N carrier frequencies as

  $\rightarrow f_b,\ f_b + (W/N),\ f_b + 2(W/N),\ \ldots$

  $\rightarrow$ ex.: $N = 100$

  $\rightarrow$ 2.4 GHz, 2.401 GHz, 2.402 GHz, . . . , 2.499 GHz

Can we squeeze in arbitrarily many carrier frequencies?

→ in principle, yes; in practice, no

Example: indoor wireless signal propagation

→ time duration of single bit: called symbol period τ

→ cannot be too short due to multi-path propagation which causes delay spread

→ i.e., time delayed echos may overlap with next bit transmission

→ called inter-symbol interference (ISI)

→ different from ICI (inter-channel interference)

Given symbol period τ to prevent ISI, cannot send bits faster than $\bar{f} = 1/\tau$ Hz.

$\rightarrow$ use as orthogonal frequency spacing

Hence number of carrier frequencies is

$\rightarrow N = W/\bar{f}$

Example (wireless): IEEE 802.11g WLANs

$\rightarrow$ uses OFDM

$\rightarrow$ symbol time $\tau = 3.2\ \mu$s

$\rightarrow$ part of IEEE standard

$\rightarrow \bar{f} = 1/\tau = 312.5$ kHz

$\rightarrow W = 20$ MHz, $N = W/\bar{f} = 64$

Example (wireline): ADSL

$\rightarrow$ frequency spacing influenced by noise factors

$\rightarrow$ ADSL: $\bar{f} = 4.3125$ kHz

$\rightarrow$ part of ITU G.992.1 standard

$\rightarrow$ UTP (unshielded twisted pair) copper wire

$\rightarrow$ frequency band: 0–1.104 MHz

$\rightarrow$ $N = W/\bar{f} = 256$

Shannon showed that there is a fundamental limitation to reliable data transmission.

- ther wider the bandwidth (Hz) the higher the reliable throughput
- the noisier the channel, the smaller the reliable throughput

  $\rightarrow$ overhead spent dealing with corrupted bits

Channel Coding Theorem (Shannon): Given bandwidth W, signal power P_S, noise power P_N, channel subject to white noise,

$$C \;=\; W \log\left(1 + \frac{P_S}{P_N}\right) \text{ bps.}$$

$\rightarrow P_S/P_N$: signal-to-noise ratio (SNR)

Implications for networking:

- Increase bandwidth W (Hz) to proportionally increase reliable throughput

  → e.g., FDM, OFDM

  → best possible way

  → wireless bandwidth: scarce resource

- Power control (e.g., handheld wireless devices)

  → trade-off w.r.t. battery power

  → trade-off w.r.t. multi-user interference: doesn't work if everyone increases power

  → signal-to-interference ratio (SIR)

Signal-to-noise ratio (SNR) is expressed as

$$\text{dB} = 10 \log_{10}(P_S/P_N).$$

Ex.: Assuming a decibel level of 30, what is the channel capacity of a telephone line?

First, $W = 3000$ Hz, $P_S/P_N = 1000$. Using Channel Coding Theorem,

$$C = 3000 \log 1001 \approx 30 \text{ Kbps}.$$

$\longrightarrow$ compare against 28.8 Kbps modems

$\longrightarrow$ what about 56 Kbps modems?

$\longrightarrow$ xDSL lines?

Nyquist's sampling criterion:

→ modern communication: mainly for digitizing analog audio (music and voice)

→ key issue: digitizing time

→ digitizing amplitude: less critical due to log-response of auditory system

Sampling Theorem (Nyquist): Given continuous bandlimited signal $s(t)$ with bandwidth W (Hz), $s(t)$ can be reconstructed from its samples if

$$\nu > 2W$$

where ν is the sampling rate.

$\longrightarrow$ ν: samples per second

Ex.: human auditory system

$\rightarrow$ sensitivity: 20 Hz–20 KHz range (roughly 20 KHz)

$\rightarrow$ voice: 300 Hz–3.3 KHz (roughly 4 KHz)

T1 TDM line: 1.544 Mbps

$\rightarrow$ frame size 193 (24 users, 8 bits-per-user, 1 preamble bit)

$\rightarrow$ 8000 samples per second

$\rightarrow$ $193 \times 8000 = 1.544$ Mbps

CD quality audio: 44100 samples per second

$\rightarrow$ also denoted Hz (44.1 KHz)

DVD quality audio: 96 samples per second (and higher)