Telephony hardware and technologies – CIS218

POTS – Plain old telephone system

T1 line – 1.544mbit/sec full duplex

- Full duplex = you can transmit and receive at the same time
- 24 channels at 64 kbit/sec
- Crystal clear call quality
- Requires hardware that can talk T1
- Normally delivered on a single pair of copper using HDSL2, or on two pairs of copper using HDSL
- Two forms used in telephony applications
  - Channelized – using robbed bit for the singling (56kbit effective)
  - ISDN PRI – using a dedicated D channel (64kbit effective)
- Depending on the market, ISDN PRI T-1 tends to run between $300 and $1500/month
  - $500/month = $22 / per channels

Key System:

- Lines
  - Telephone number and “channel” are one and the same
  - Put someone on hold on “line 2” and yell for someone else to pick up “line 2”

PBX:

- Trunks: Inbound or outbound phone channels
- DIDs: Direct inward dialing – Phone numbers

When you buy dids, you buy blocks in most cases, and buy extra
For example, one company I worked for bought two entire central office prefixes of 10,000 numbers each: 520-872-XXXX 520-873-XXXX

Trunks:

FXO – Single analog line from the telephone company

- One number / one line relationship
- Echo issues due to single pair configuration

ISDN-BRI – Two digital lines from the telephone company on a single pair

- Little to no echo issues
- 128Kbit bonded internet/data capability
- 2B+D (Bearer channel 64kbit = digital phone line), (Data channel 16kbit = signaling)

T-1 RBS – T1 line with Robbed bit signaling

- 24 64Kbit channels, with 8bits robbed per channel for signaling
- 24 56Kbit usable channels, signaling is similar to analog, with tones per channel
- AMI and B8ZS signaling channels

T-1 ISDN-PRI – 23 digital lines from the telephone company on one or two pairs (two pairs for hand over to customer as most CPE uses two pairs for T1/PRI)

- Little to no echo issues
- Supports DIDs due to the ISDN “D” channel for signaling

23B+D (Bearer channel 64kbit = digital phone line), (Data channel 64kbit = signaling)
Cost: $350-800/month

SIP Trunk - Session Initiation Protocol (signaling) combined with RTSP (Real Time Streaming Protocol)

Digital channel over Internet Protocol – Normally with UDP packets
Variable latency – aka Jitter can be problematic
High latency – Normally will still work, though the users may notice it
G.711 (ulaw and alaw) – Uncompressed digital trunk 80Kbit/sec with IP overhead
G.729a – Highly compressed digital trunk 24Kbit/sec with IP overhead
GSM – Highly compressed digital trunk – 29Kbit/sec with IP overhead
G.722 – Highly compressed, HD or high fidelity codec 16KHz samples instead of the standard 8 KHz samples – 80Kbit/sec with IP overhead
IP Ports 5060 (setup) RTSP picks almost anything over 10,000

IAX2 Trunk - Inter Asterisk Protocol

Uses a single port, 4569, and is much more NAT friendly
Digital channel over Internet Protocol – Normally with UDP packets
Variable latency – aka Jitter can be problematic
High latency – Normally will still work, though the users may notice it
G.711 – Uncompressed digital trunk 64Kbit/sec plus combined overhead
G.729a – Highly compressed digital trunk 8Kbit/sec plus combined overhead
GSM – Highly compressed digital trunk – 13Kbit/sec plus combined overhead
G.722 – Highly compressed, HD or high fidelity codec 16KHz samples instead of the standard 8 KHz samples – 64Kbit/sec plus combined overhead

VoIP over Public Internet:

SIP/IAX2

Variable latency that you do not have any control over
Some ISPs block traffic (more common in Mexico and Latin America)
Not encrypted by default
Best for tying asterisk systems together due to reduced overhead
Not many telephone companies support IAX2

VoIP over private line:

SIP/IAX2

Provider controlled line, with latency and up-time guarantees
Not encrypted, but it's direct between you and your provider, limiting who can be in the middle

Channel Bank

Takes in either RBS or PRI T1 (RBS is fine), and provides 24 analog ports

Depending on the model can be configured for all FXO all FXS, or a mix, and some can do data break out as well.
FXO = Central Office
FXS = Telephone station

http://www.asteriskguru.com/tools/bandwidth_calculator.php