Telephony, Telephone, Voice, and Voice over IP technologies.

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 pickup "line 2"

PBX:
Trunks: Inbound or outbound phone channels
DIDs: Direct inward dialing – Phone numbers
Carondelet – 520-872-XXXX 520-873-XXXX
When you buy dids, you buy blocks in most cases, and buy extra

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
Supports DIDs due to the ISDN "D" channel for signaling
128Kbit bonded internet/data capability
2B+D (Barer 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)
Little to no echo issues
Supports DIDs due to the ISDN "D" channel for signaling
23B+D (Barer 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 – 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
Best for tieing asterisk systems together due to reduced overhead
Not many telephone companies support IAX2

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

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