ANALOG TELEPHONY

Pair

- Tip—ground (positive, relatively)
- Ring—“Battery” negative = -48 volts DC

On Hook—Tip and Ring are separated at the phone

Loop Start Signaling—Typical in homes. When the phone is on-hook, tip & ring are separated; no current. When the receiver is lifted, tip & ring are connected, and 48volts flows. This is exactly the same whether a call is being placed or answered.

Glare—a problem with loop start signaling. A person picks up the phone to dial out at the moment a call comes in. They are connected without the phone ever ringing. In a business key-system, an extension might be given the PSTN line to make an outgoing call and get an incoming call (meant for someone else) instead.

Ground Start Signaling—Originated with pay phones. No dial tone until tip and ring are momentarily grounded (e.g. by a passing coin), telling the phone company that someone wants to make an outgoing call. Used by most PBX (Private Branch Exchange) connections to the telco to avoid glare.

Other Signaling

- Supervisory Signaling—on hook, off hook, ringing
- Informational Signaling—dial tone, busy, rignback
- Address Signaling—DTMF (Dual-Tone MultiFrequency) aka touch tone, pulse (rotary phone)

Analog Shortcomings

- Signal Degradation—signal weakens over distance. Repeaters can amplify it, but they also amplify noise.
- Lots of Wires—each signal / conversation needs a separate pair

DIGITAL

Digital allows multiplexing multiple calls onto one pair and can be amplified by repeaters without raising the noise floor.

TDM (Time Division Multiplexing)—multiple digital phone connections on the same 4-wire connection using time slots.

DS0 (Digital Signal Zero)—one 64Kbps time slot; one phone call

T1—24 x DS0. USA, Canada, Japan. Common business connection to a CO (PSTN Central Office)

E1—30 x DS0—The rest of the world

Frame—How TDM is implemented—One byte from each DSO plus a framing bit.

CAS (Channel Associated Signaling)—Signaling data embedded in the channel. AKA Robbed Bit Signaling (RBS). Borrowed bits only transmit data relevant to their own channel.

- Within each channel, use eighth bit of every sixth sample (when talking about the entire T1/E1, this might be expressed as the 6th frame)

CCS (Common Channel Signaling)—Separate dedicated channel for signaling—"out-of-band."

- Q.931 is most popular, used for ISDN
- More popular for voice channels—More flexibility in choosing a signal protocol and communicating proprietary data between PBXs, better security by keeping signaling separated, higher voice quality by not stealing a bit
- T1 uses the 24th time slot; E1 uses the 17th

PSTN

PSTN (Publicly Switched Telephone Network)

Local Loop—Telco CO (Central Office) to customer endpoint

CO Switch—provides signaling, digit collection, call setup, routing, and teardown to local loop

Trunk—connects switches, telco or private (PBX).

Private Switch—saves on connections to CO

Digital Telephone—connects to PBX, converting voice to digital internally

PBX (Private Branch eXchange)—In-house phone system for large companies, composed of line cards (for phones), trunk cards (connections to PSTN or other PBX), and a control complex

Key systems—fewer users and a "shared line" feel

PSTN Connections—as a business grows, individual lines can be consolidated in digital T1 lines.

SS7—Voice signaling protocol used within the PSTN. CCS (out-of-band). Details out of scope.

- Performs lookup to find call destination device in the world
- Routes Call
- Provides "ringback" to the calling device

E.164—Number plan for phone numbers. 15 digits max. Defined by the ITU (International Telecommunication Union)

NANP (North American Numbering Plan)—More specific than E.164, but compliant, as shown.

E.164	Country Code	National Destination Code		Subscriber Number
NANP	Country Code	Area Code	Central Office Code	Station Code
Example	1	(602)	555 -	1212

Benefits

- Communication Costs—no toll charges on data lines already in place
- Cabling Costs—In new construction one Ethernet line for data and phone
- Seamless Internal telephone network—branch offices, telecommuters, dialed as if internal.
- Moves, Adds, Changes (MAC) essentially free as phones auto-configure selves
- Software IP phones (headset on computer) can even integrate with other software apps
- Unified messaging—voicemail, e-mail, fax all go to e-mail
- Forward extension to ring multiple devices at once before voicemail
- Feature-rich call integration with computer software, voice, video, etc.
- Open Standards

DIGITAL AUDIO

Use	Frequencies
Human Hearing	20 – 20,000
Speech	200 – 9,000 (according to book)
Telephone	300-3,400

Note: Actually, men speak at 85-180 Hz, average 125 (B_2), Women 165-255 Hz, average 200 (G_3), but Cisco uses the numbers above. Children speek 250-400 Hz (B_3 to G_4).

Dr. Harry Nyquist (February 7, 1889 – April 4, 1976)—Found that 2 samples per highest frequency allow reconstruction.

Quantization—Amplitudes measured at fixed frequency and converted to an 8-bit number. Measurement is non-linear, concentrating bits at low amplitudes (companding). The effect is as if the loudness was squashed prior to measurement and expanded at the other end to ensure that even quiet speech uses plenty of bits (resolution/detail).

- 8-bits per sample (sign + 7-bit amplitude)
- 8,000 samples per second (total 64 Kbits/second)

All Cisco phones do G.711 (uncompressed 8-bit 8KHz sampling) and G.729 compressed. G.711 is the common ground for all IP phones.

G.711 codec is uncompressed (but companded)—Two versions

- μ-law (USA & Japan) version 1st bit 1 = negative number
- A-law (everywhere else) version has 1 = positive number (perhaps an unsigned number with a wave-centering offset). Conversion swaps 1s for 0s throughout.

G.729—Compresses to 8 Kb/s per call. Uses a code book to say “keep playing this sound.” G.711 and G.729 are supported by all Cisco phones.

G.722—Cisco's new default phone codec for phones—wider frequency range, but same 64 kbps.

Mean Opinion Score (MOS)—Listener rates the phrase "Nowadays a chicken leg is a rare dish" 1-5.

Operational Status	Bandwidth	MOS
G.711 (uncompressed)	64 Kbps	4.1
G.722	64 Kbps	4.2 or so
G.729	8 Kbps	3.92
G.729a	8 Kbps (but more processor efficient)	3.7

DSP SIMMS

PVDM (Packet Voice DSP Modules)—resemble a memory SIMM and can be added to motherboard slots in an ISR (router) or to a network module that in turn supports voice cards. PVDMs handle analog/digital conversion for gateway connections to the PSTN and transcoding between different codecs. PVDM3 modules are more powerful than PVDM2 and can process video as well as voice.

Codec Complexity—DSPs can handle twice as many medium-complexity codec calls as high. Newer chips are more powerful than older. Use Cisco's DSP calculator to figure out what you really need.

Medium-Complexity	High Complexity
G.711 (uncompressed)	iLBC (Internet Low Bitrate Codec)
G.729a, G.729ab	G.729, G.729b
G.726	G.723
	G.728

PROTOCOLS

RTP (Real-time Transport Protocol)—L4 atop UDP (also L4). UDP provides port numbers (session multiplexing) and checksums for header correctness. RTP provides

- RTP Type—video or voice
- Sequence Numbers
- Time stamps—buffering for jitter removal

When two devices connect, RTP chooses a random even port number (16,384 – 32,767) for each stream (each direction) for the duration of the call. Two needed if it's a conversation.

RTCP (Real-time Transport Control Protocol)—statistics like packet count, delay, loss, and jitter at least once every 5 seconds

- Uses the odd-numbered port immediately following the RTP connection
- Not nearly as important as RTP for QoS purposes