1 Background

Analog

Tip	Ground (positive, relatively)
Ring	"Battery" -48 VDC
On-hook	Tip & Ring separated at phone
Home-style, where 48volts flows on call & answer	Loop-Start Signaling
PBX-style, no dial tone until tip & ring momentarily shorted to zero	Ground-Start Signaling
Problem with home-style, receive when lift to call	Glare

Digital

TDM	Time Division Multiplexing
DS0	one 64 Kbps time slot = one phone call
T1	24 x DS0
E1	30 x DS0
CAS	Channel Associated Signaling (RBS) (in-band)
CCS	Common-Channel Signaling (out-of-band)
Q.931	Most popular CCS, used for ISDN
RBS	Robbed-Bit Signaling (CAS)
Time-slot used in T1 for non-audio overhead	24th of 24
Time-slot used in E1 for non-audio overhead	17th of 30
Which bits used for in-band	8th bit of every 6th sample
SS7	Signaling protocol used by PSTN. (out-of-band). Performs destination lookup from number, Routes call, provides "ringback" to calling device
E.164	ITU plan for phone numbers (15 digits max)
NANP	North American Numbering Plan (11-digit)
MAC	Moves Adds Changes (phones within a company)
Nyquist	(Dr. Harry) 2 samples per highest frequency
Quantization	Turning amplitudes into numbers
MOS	Mean Opinion Score ("Nowadays a chicken leg is a rare dish")
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol

Frequencies

Hearing	Speech	Phone
20 - 20k	200 - 9000	300 - 3400

Uncompressed Digital Phone Audio

Bits per Sample	Samples per Second	Total Bits / Second	Highest Frequency
8 (sign + 7)	8000	64 Kbps	4 KHz

Codecs

Codec	Bandwidth	MOS
G.711	64 Kbps (companding only)	4.1
G.729	8 Kbps	3.92
G.729a	8 Kbps (more processor efficient)	3.7

Codec Complexity	**High**	**Medium**
G.711		X
G.729	X	
G.729a		X
G.729ab		X
G.729b	X	
iLBC	X	

	USA & Japan	**MSB 1 = negative**
G.711 μ-law	X	X
G.711 A-law		

VoIP Benefits
No toll charges
Cabling Costs (share data & phone)
Seamless dialing to branch offices
MACs free
Softphones can integrate w/ other software
Unified Messaging (voice/fax to e-mail)
Ring multiple devices at once before voicemail

Call Signaling and Media Flows

Purpose of RTCP (contents carried)	Call Statistics every 5 seconds—delay, loss, jitter
Purpose of RTP (contents carried)	Call audio
RTP Port	UDP random even 16,384-32,767 each direction
RTCP Port	UDP odd port RTP + 1
RTP OSI Layer	4 atop UDP (also 4)
RTP Provides:	RTP Type (video / voice)
	Sequence #s
	Time Stamps—buffering for jitter removal