TCOM 631:
Voice over IP
George Mason University

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Agenda

Pre-VoIP - Telephony 101

Intro to voice signal and digitization

Brief intro to digital signal transmission

Legacy telephony concepts and architectures

Why VoIP

High-level VoIP architectures and connection strategies

Pre-VoIP Telephony 101

Fundamentals of early telephony system exist today!

Hello, Hello,.....

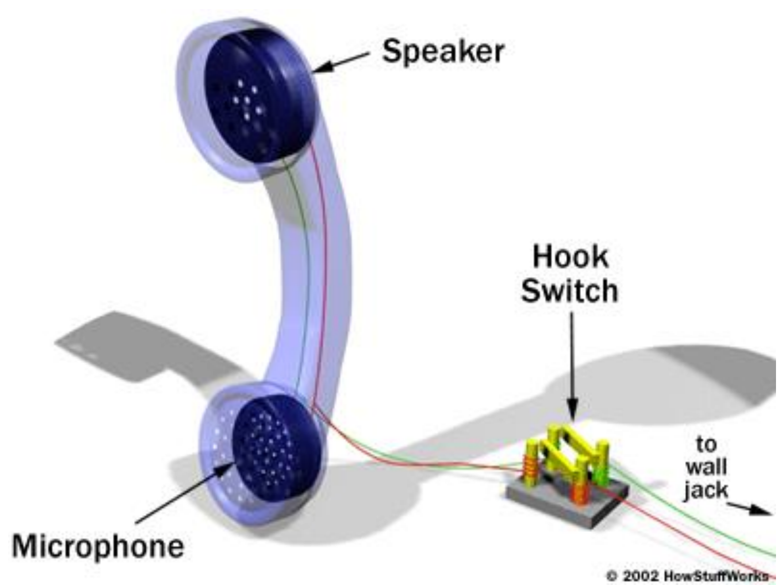
- The telephone works by converting acoustic energy into electrical energy
- It turns the sound waves of the speaker's voice into a varying electric current which is sent along a wire and is then turned back into sound waves
- Alexander Graham Bell (Scottish-born scientist, inventor, engineer) is usually credited with the invention of the telephone which was patented in 1876
  - At least ten men before him had the idea of the telephone and two of them produced a practical telephone
  - Philip Reis (German scientist and inventor) made a telephone in 1863 but did not take out a patent
  - Elisha Gray (American electrical engineer) also invented a telephone but was beaten to the patent office by Bell by a few hours

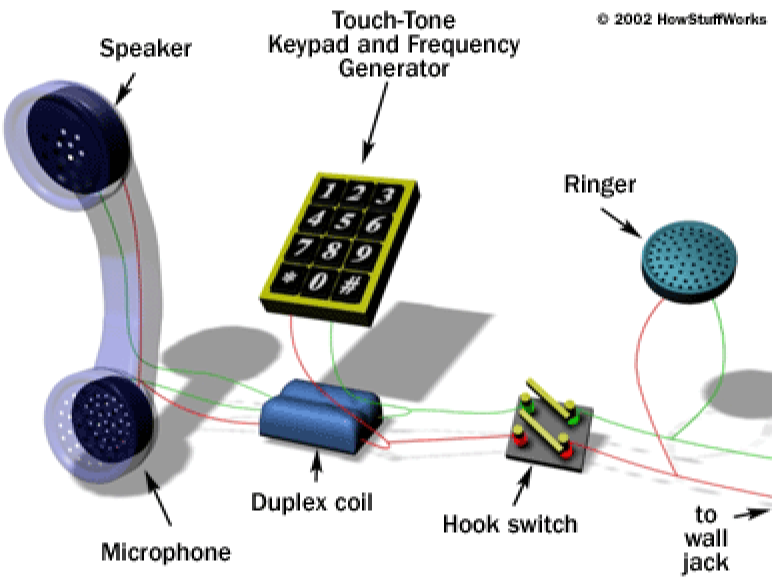
Telephony Equipment

- Telephone sets (analog or digital)
- Telephone switches
  - Key systems
  - PBXs
  - Central Office
  - Tandem
  - Etc.

Telephone Set

- A telephone typically consists of the following components:
  - Handset containing a Microphone and Speaker
  - Switch hook, which is a lever that is depressed when the handset is resting in its cradle
  - Two-wire to four-wire converter to provide conversion between the four-wire handset and the two-wire local loop
  - Duplex coil to block the sound of your own voice from leacking back
  - Keypad (either rotary or touch-tone)
  - Ringer





Microphone and Speaker

- Microphone turns the sound energy from voice into electrical energy
  - The microphone contains a flexible piece of plastic called a diaphragm with an iron coil attached to it and a nearby magnet
  - When you speak the sound energy in your voice makes the diaphragm vibrate, moving the coil nearer to or further from the magnet
  - This generates an electric current in the coil that corresponds to the sound of your voice
  - If you talk loud, a big current is generated; if you talk softly, the current is smaller

- The loudspeaker in a phone works in the opposite way
  - It takes an incoming electrical current and uses magnetism to convert the electrical energy back into sound energy you can hear

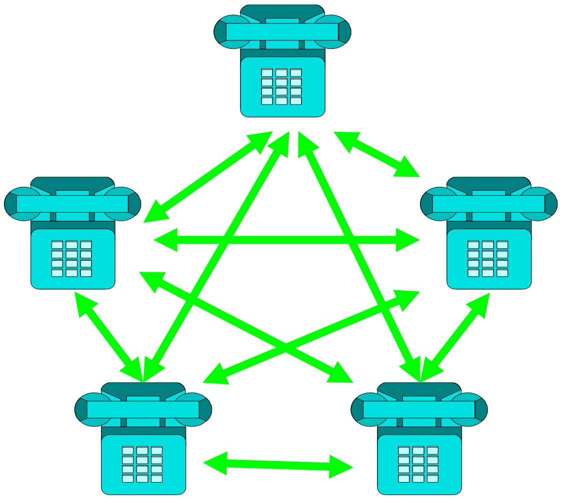
- An electret microphone (dielectric material) is the most typical in consumer electronics today (smart phones, etc.)
  - Many different kinds: https://en.wikipedia.org/wiki/Microphone

Basic Operation

- A typical telephone does the following:
  - Requests service from the network
  - Performs dialing functions
  - Performs a notification function (it rings)
  - Provides answer and disconnect supervision
  - Converts outgoing speech to electrical signals, and vice versa

The N^2 Consideration

- For N users to be fully connected directly
- Requires N(N – 1)/2 connections
- Requires too much space for cables
- Inefficient & costly since connections not always on

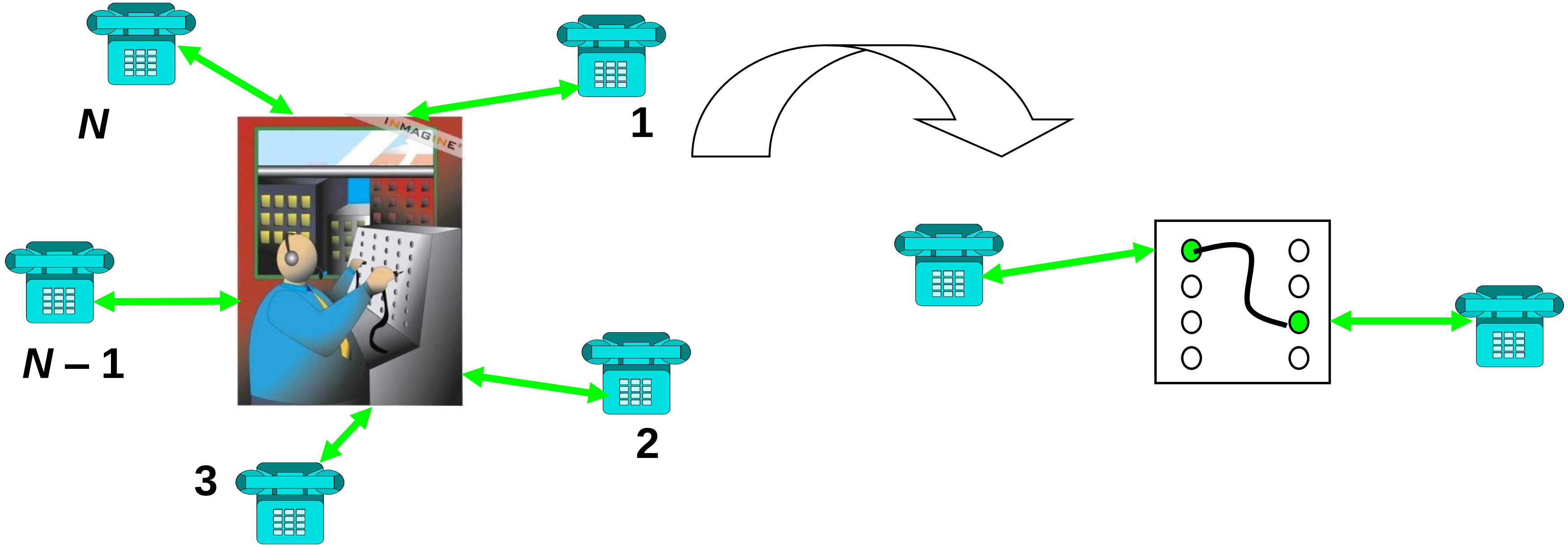


$N = 100000$

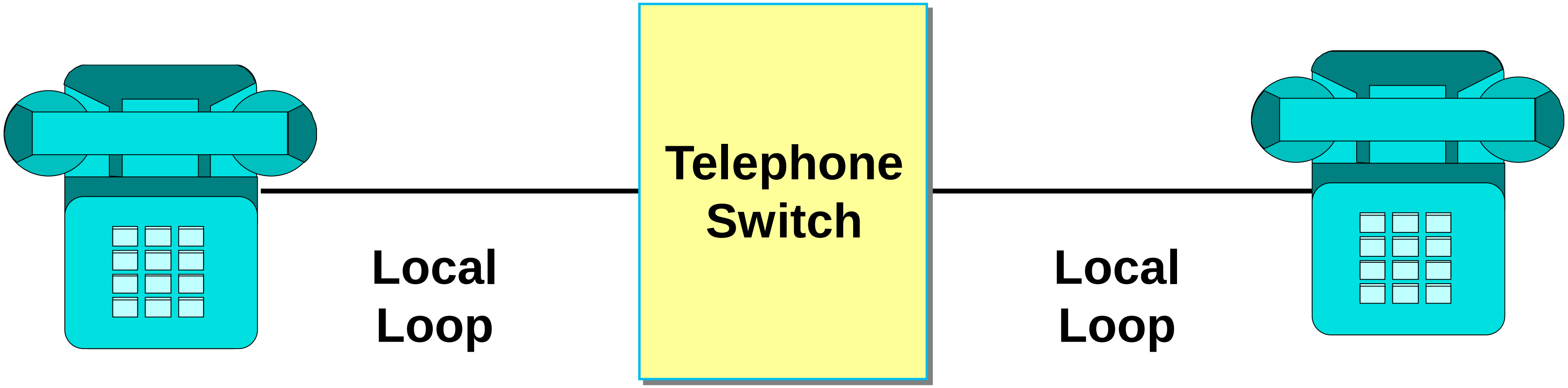
$N(N - 1)/2 = 4{,}999{,}950{,}000$

Circuit Switching

- Centralized switching power was needed
  - Patchcord panel switch invented in 1877
- Operators connect users on demand
  - Establish circuit to allow electrical current to flow from inlet to outlet
- Only N connections required to central office (CO)
- Major switching revolution to follow…

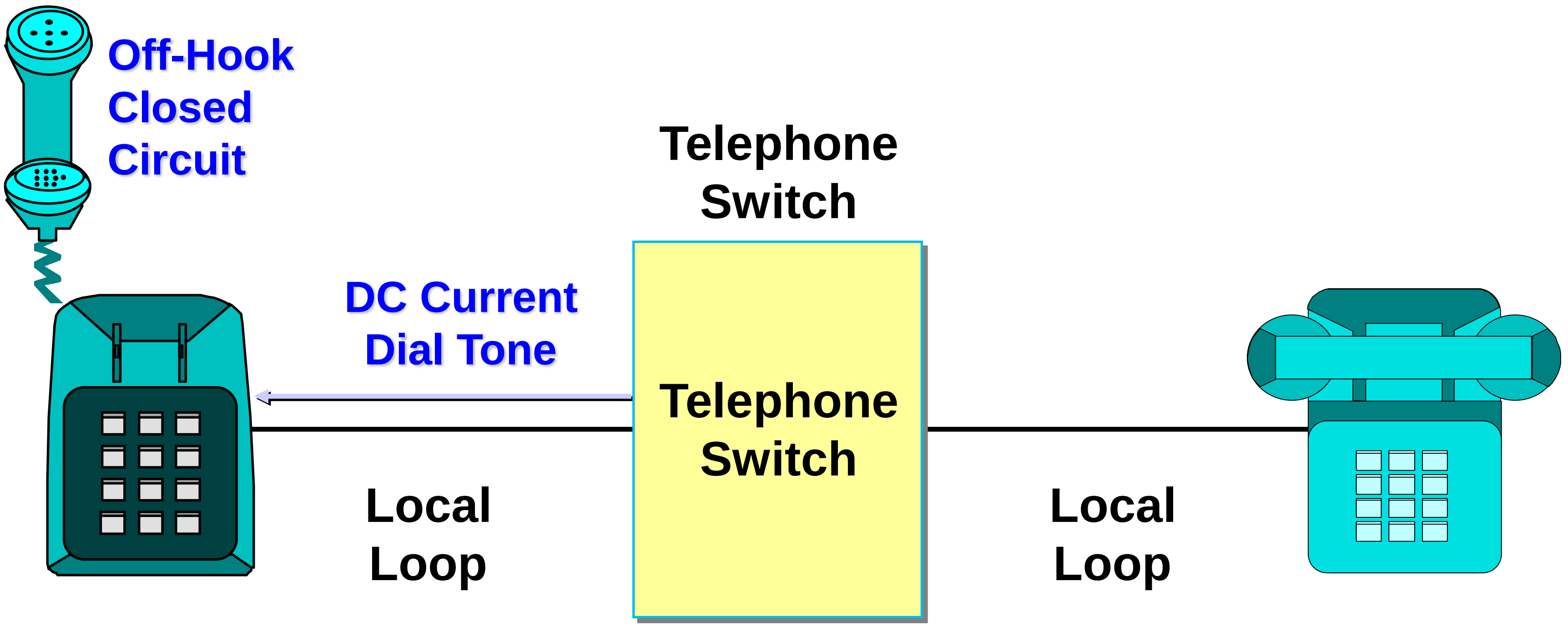


Basic Call Progress: On-Hook

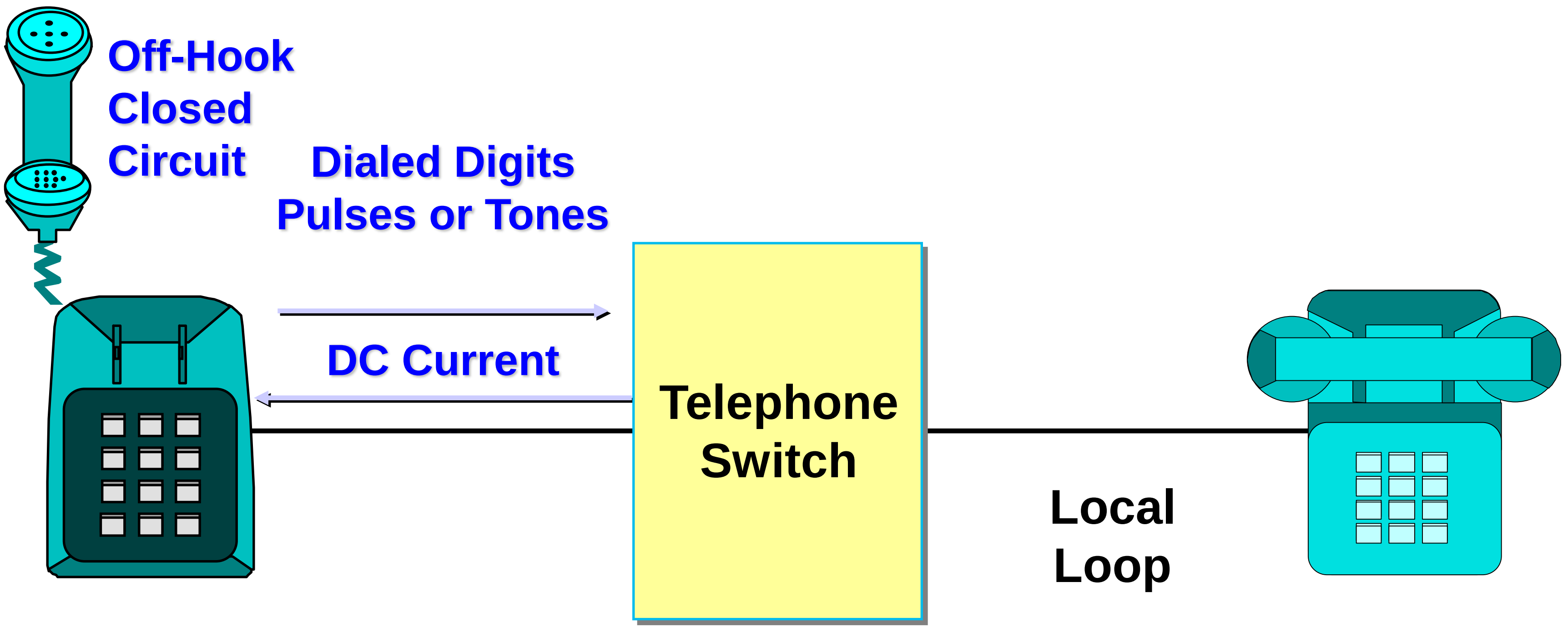


-48 DC Voltage
DC Open Circuit
No Current Flow

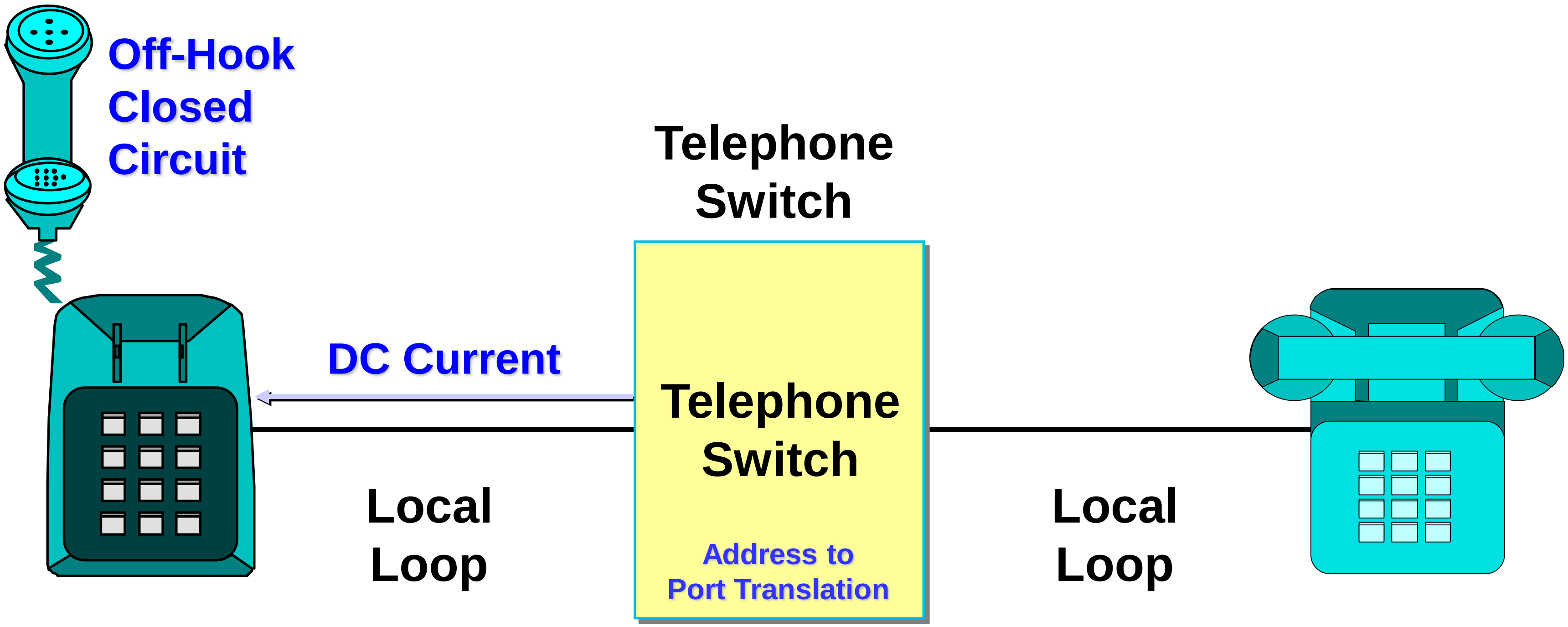
Basic Call Progress: Off-Hook



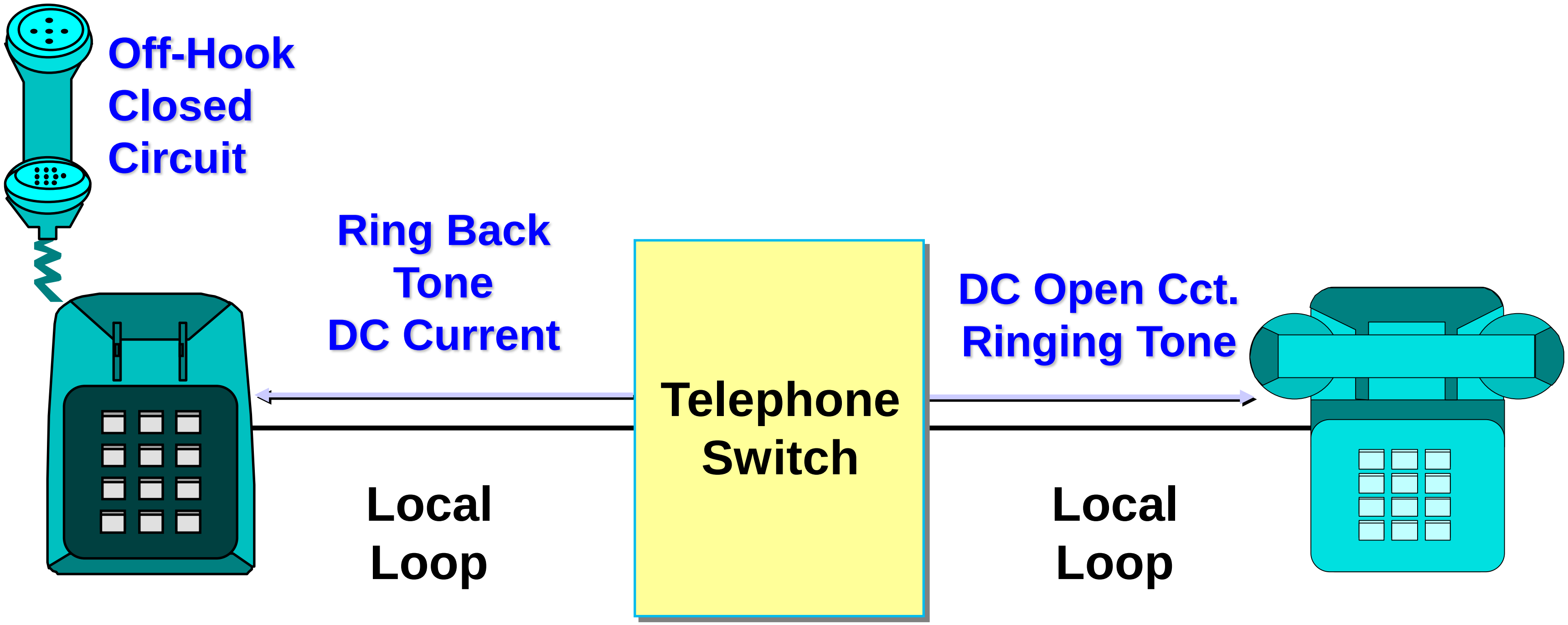
Basic Call Progress: Dialing



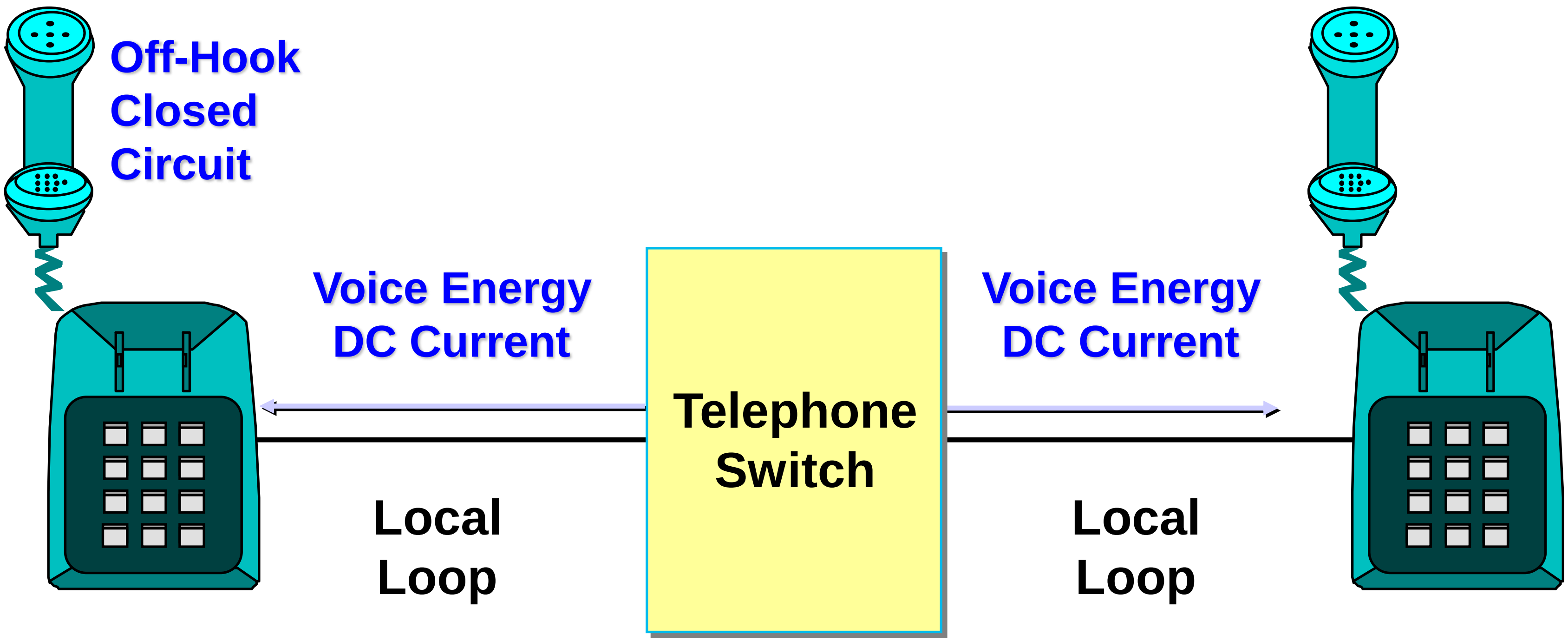
Basic Call Progress: Switching



Basic Call Progress: Ringing



Basic Call Progress: Talking



Analog Telephony - Signaling

- The purpose of signaling in a voice network is to establish a connection
  - Flashing light and ringing devices to alert the called party of incoming call
  - Called party information to operator to establish calls
  - Control information that implies change of status in the network, along the signaled path (includes call records allocated for the call (timeslots, senders and receivers, memory, processes, etc.)
  - Signals can be impulses, voice band tones, binary signals or messages transported in a packet network

- Three forms of signaling:
  - Access Signaling
  - Station Loop Signaling
  - Address Signaling

Lecture 1: Introduction

A little about Voice

- Sound is a disturbance of mechanical energy that propagates as a wave

- Sound waves, like other waves, are caracterized by:
  - *Frequency*: Represents the number of periods in a second and is measured in *hertz* (Hz) or *cycles per second*
    Human hearing frequency range: 20Hz to 20kHz (audio)
  - *Amplitude*: The measure of displacement of the air pressure wave from its mean
  - *Wavelength*: The distance between repeating units of the propagation wave

A little about Voice

- A voice frequency is one of the frequencies, within part of the audio range, that is used for the transmission of human speech
- Usable (intelligent) voice frequency ranges from approximately 300 Hz to 3400 Hz and it is called voice band
- Voice channel has a range of 0 – 4kHz (narrowband coding). Area between 3.4 and 4 kHz is used for system control

Voice Stream Information

- A real-time voice signal is digitized and transmitted as it is produced at the constant rate between sender and receiver
- Speech signal level varies continuously in time (quasi-periodic)
- Speech is slow-varying signal
  - From sample to sample there is a **significant amount of correlation**!

Normal speech consists of talkspurts, which typical last a few hundred milliseconds and silence periods, which occur within a spoke word and between words.

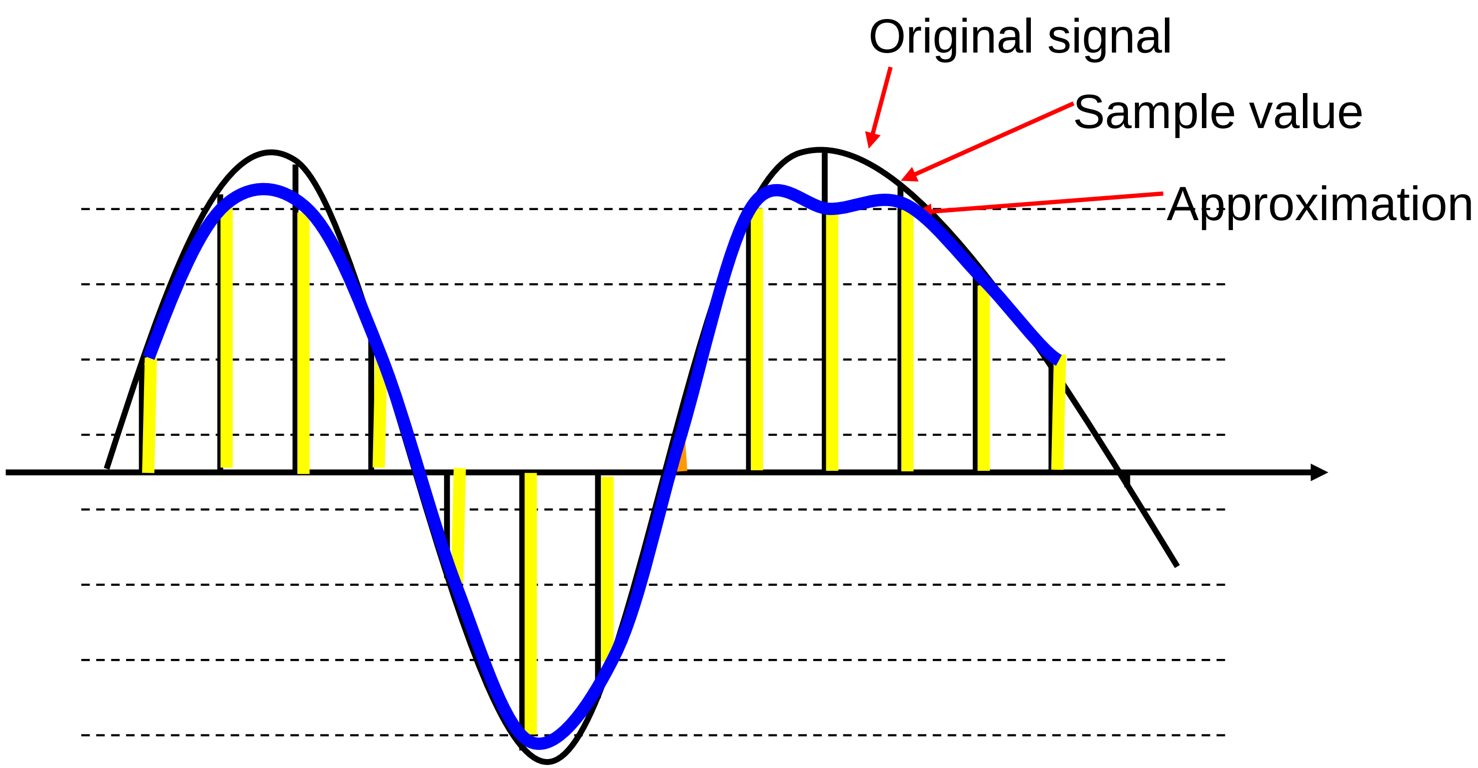
Digitization of Voice Analog Signal

- Why voice digitization?
  - Digital computation is much easier
  - Easier noise elimination
  - Ensures better quality
  - Immune to the imperfections
  - Provides higher capacity
  - Supports longer transmission distance
- Different voice digitization techniques exist
  - In this lecture we will talk about Pulse Code Modulation

Pulse Code Modulation

- PCM is invented by Alec H. Reeves (English inventor) in 1938
- Interesting fact: PCM bandwidth grater then that required by the original analogue signal
- Two principles of digitization: Sampling and Quantization
  - Sample analog signal in time and amplitude
  - Find closest approximation
- Sampling: Sampling refers to taking “snapshots” at discrete points in time. Discretizes the continuous time
- Quantization: Quantization refers to the approximation of the continuous signal value by a number, usually an integer, and almost always a number in the range of 0 to 2^N - 1 where N is called the bit depth. For example, 8-bit quantization providesinto 256 descrete levels. 16 bit gives you 65536 levels. Lower the quantization, lower the quality of the sound
  - Reduces the infinite range of the sampled amplitudes to a finite set of possibilities
  - Linear vs. Non-Linear quantization (more to come)

Digitization of Voice Analog Signal



Methodology

Voice Sampling

- Harry Nyquist (1889-1976)
  - Born in Sweden, moved to USA in 1907, AT&T>>>Bell Labs
  - Dr. Nyquist and Dr. Claude Shannon are responsible for virtually all the theoretical advances in modern telecommunications

    http://en.wikipedia.org/wiki/Harry_Nyquist

- 4 kHz assumes conversational speech, not singing or CD-quality audio

- Sampling twice per cycle allows us to reconstruct the signal – which is what we want to do to rebuild the voice signal at the receiving end

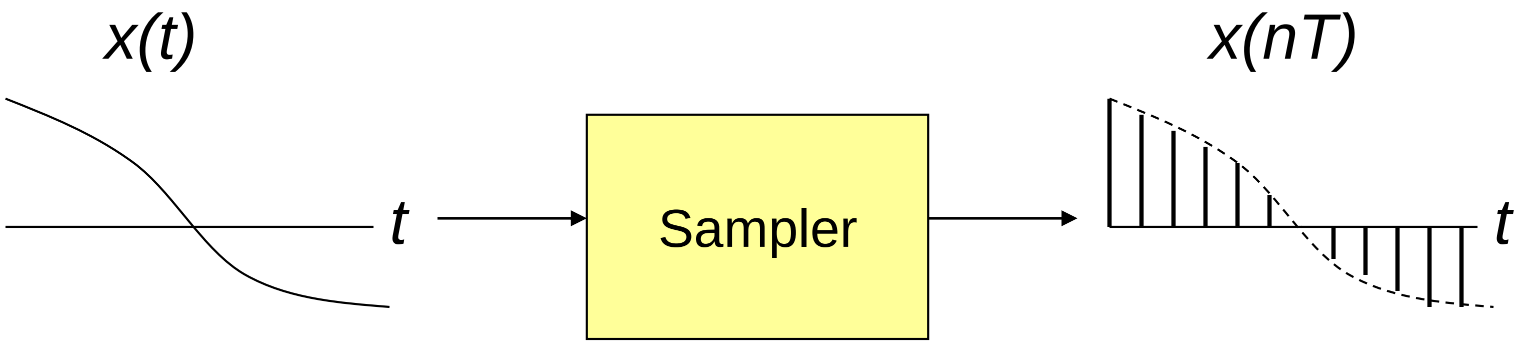
Limit of human hearing: ~22KHz?

CD sample rate: 44,100 samples/sec

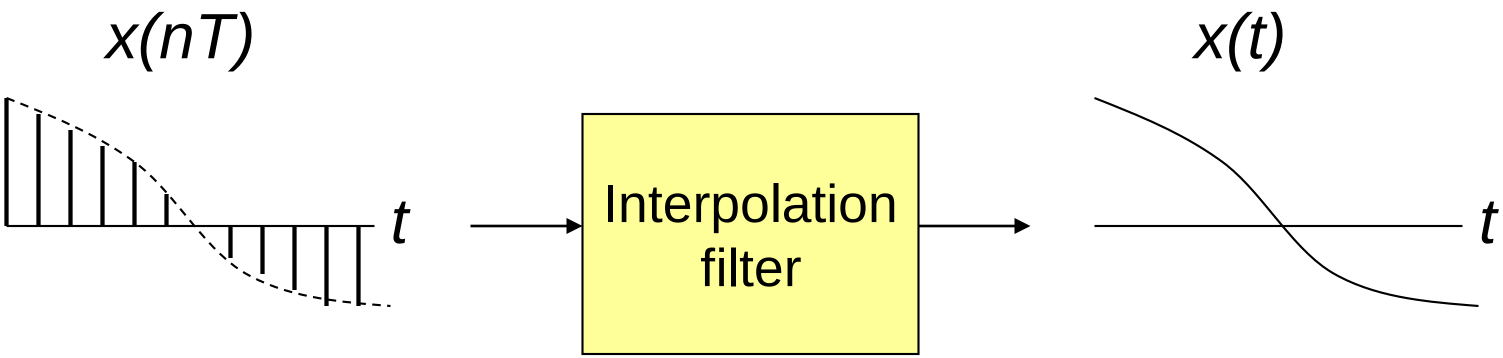
Sampling Theorem

Perfect reconstruction if sampling rate 1/T > 2 * maximum signal frequency

(a)



(b)

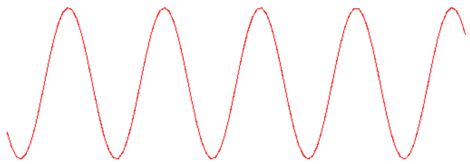


n – index of the sample

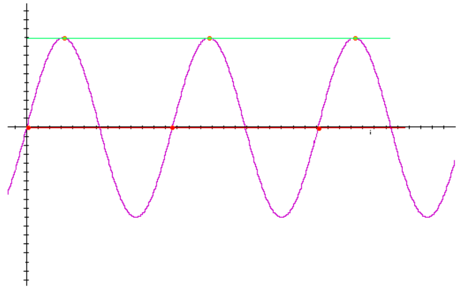
T – sampling period

What happens to all those higher frequencies you can't sample?
They add noise to the sampled data at lower frequencies ALIASING NOISE

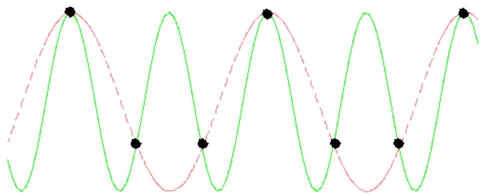
Sampling



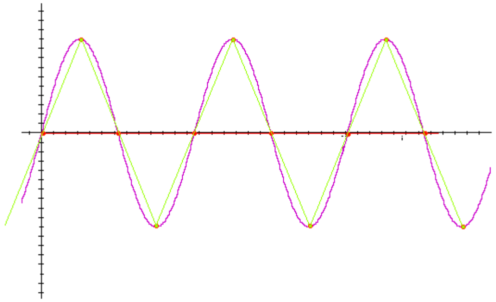
Lets consider a sign wave to be sampled (e.g. voice signal)



If you sample at 1 time per cycle, you can think it is a constant



Sampling 1.5 times each cycle appears as a low frequency sine signal



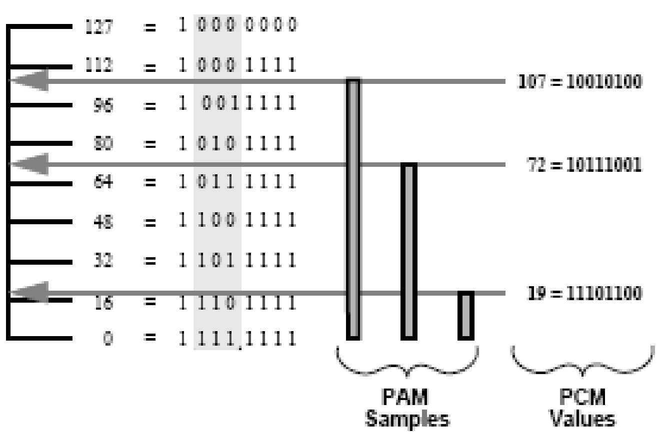
Nyquist Theorem: **For Lossless digitization, the sampling rate should be at least twice the maximum frequency responses**

Quantization

- Quantization can be *Uniform, Logarithmic, Adoptive, Differential or Vector*
- The distance between the finite set of amplitude levels is called quantizer step size and is usually represented by delta
- Each discrete amplitude level x_i is represented by a codeword c(n) for transmission purpose
  - This codeword indicates to the de-quantizer, which is usually at the receiver, which discrete amplitude is to be used
- The main aim of a specific quantizer is to match the input signal characteristics both in terms of its dynamic range and probability density function

Quantization

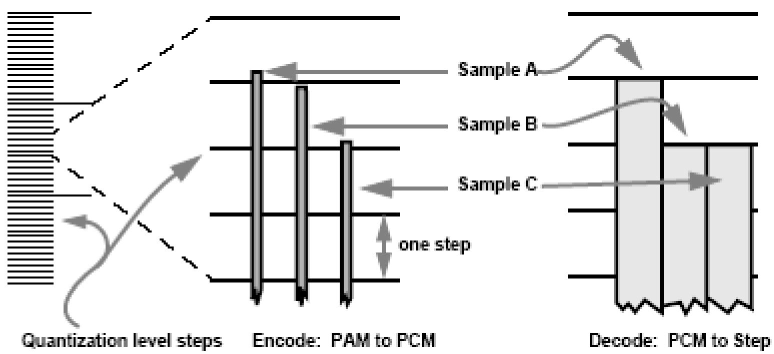
- How many bits do we need to accurately represent the range we're sampling?
  - For the 0 – 4kHz range, 8 bits is enough
- If eight bits are allowed for the PCM sample, this gives a total of 256 possible values
- PCM assigns these 256 possible values as 127 positive and 127 negative encoding levels plus the zero-amplitude level

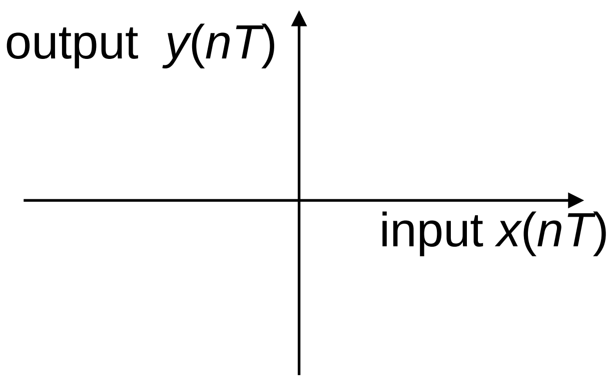


- These values translate into binary codes which become the corresponding PCM values

Quantization Error

- Each sample is a digitized estimate
  - if the analog value is 5.3, the closest we can come is 5
  - all we'll get on the other end is 5
  - the 0.3 is lost forever
- Something is lost in the sampling: *Quantization Noise* or *Quantization Error*





Quantization error (noise):
$e(n) = x(nT) - y(nT)$
$-\Delta/2 \le e(n) \le \Delta/2$ for rounding quant.,
or $0 \le e(n) \le \Delta$ for truncation quant.

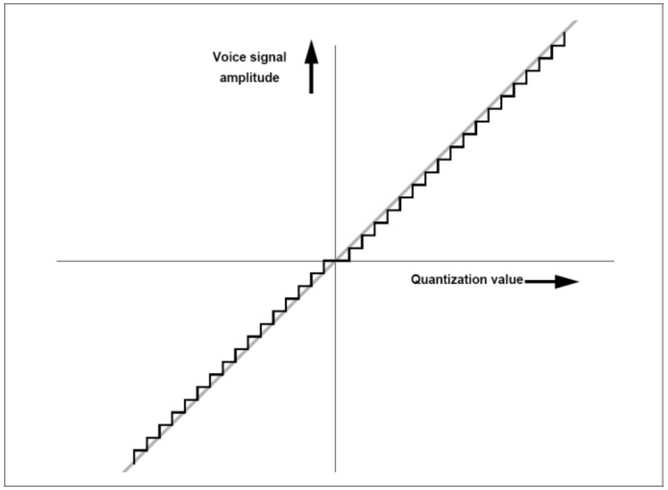
Statistically quantization noise is stationary and is unrelated to the input signal. It is also considered uniformly distributed.

Uniform Quantization

- All of the quantizer intervals are of the same width
- It can be defined by two parameters
  - Number of quantizer levels; $N_{ql} = 2^B$
  - Quantizer step size; $\Delta = X_i - X_{i-1}$
- Total mean square error for uniform quantizer: $E^2 = \frac{\Delta^2}{12}$
- Signal to Noise Ratio (SNR) formula is useful to determine number of bits needed in quantizer for the certain signal to quantization noise ratio and in estimating the performance of u uniform quantizer for a given bit rate
  - SQNR = 6.02B dB (for i.e. 8-bit codeword SQNR = 6.02*8 = 48.16 dB
  - The SQNR increases approximately 6dB for each bit

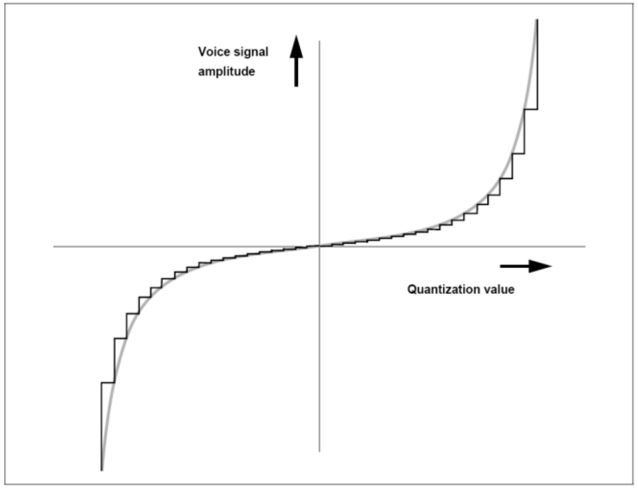
Quantization (liner vs. logarithmic)

Linear Quantization (uniform)



- The 127 quantization levels are spread evenly over the voice signal's dynamic range
- This gives loud voice signals the same degree of resolution (same step size) as soft voice signals
- Encoding an analog signal in this manner, while conceptually simplistic, does not give optimized fidelity in the reconstruction of human voice

Logarithmic Quantization



- Most of the energy in human voice is concentrated in the lower end of voice's dynamic range (no shouting – just from the boss)
- Quantization levels distributed according to a logarithmic, instead of linear, function gives finer resolution, or smaller quantization steps, at lower signal amplitudes
- 8-bit PCM in North America uses a logarithmic function called μ-law and in Europe A-law

μ-law and A-law

- In both schemas (mu-law, a-law), the signal to quantization noise performance can be very close to that of uniform quantizer
- Quntizer levels are closely spaced for small amplitudes which progressively increase as the input signal range increases
- Both give similar quality to 12-bit linear encoding

μ-law

$$F(x) = \frac{sgn(x) ln(1+\mu|x|)}{ln(1+\mu)} \qquad 0 \le |x| \le 1$$

μ is 255 in US/Japan

A-law

$$F(x) = \frac{A|x|}{ln(1+A)} \qquad 0 \le |x| < \frac{1}{A}$$

$$F(x) = \frac{sgn(x) ln(1+A|x|)}{ln(1+A)} \qquad \frac{1}{A} \le |x| \le 1$$

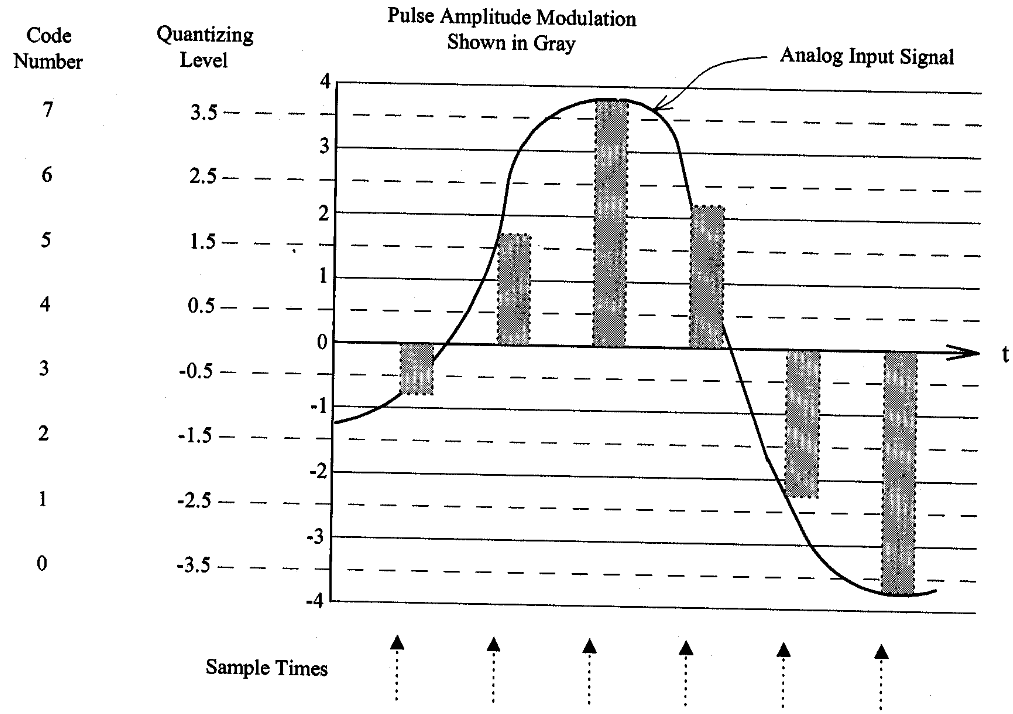
A = 87.7 in Europe

Bit Rate of Digitized Signal

- Assuming all of the discrete amplitude values in the quantizer are represented by the same number of bits B and the sampling frequency is $f_{s,}$ the channel transmission bit rate is given by
  - *$Tc = Bf_s$ bits/second*
    - Bandwidth in Hertz: how fast the signal changes
    - Higher bandwidth → more frequent samples
    - Minimum sampling frequency = 2 * bandwidth

- Given the fixed sampling frequency the only way to reduce the channel bit rate Tc is by reducing the length of the codeword c(n)
  - This creates bigger difference between the analogue and discrete amplitudes which reduces the quality of reconstructed signal
  - Various types of scalar qantizers are used in order to reduce bit rate while maintaining a good speech quality

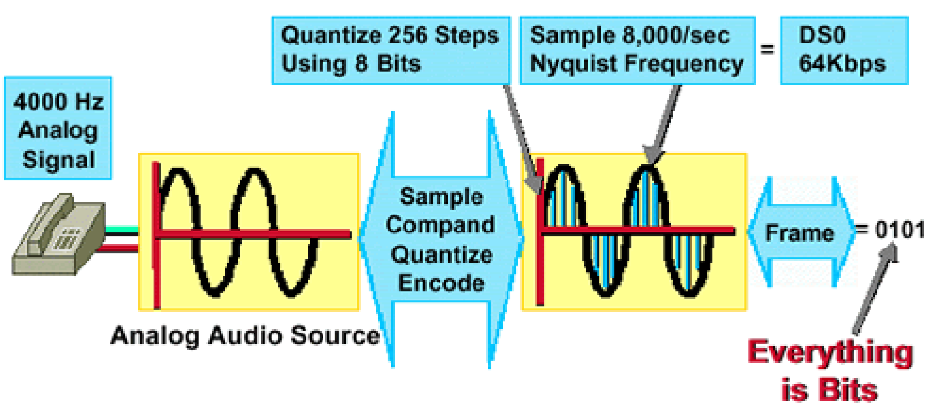
Digitization Example



Sample Value	-0.8	1.7	3.8	2.2	-2.2	-3.7
Quantized Value	-0.5	1.5	3.5	2.5	-2.5	-3.5
Code Number	3	5	7	6	1	0
Binary Code	011	101	111	110	001	000

CODEC – Coder/Decoder

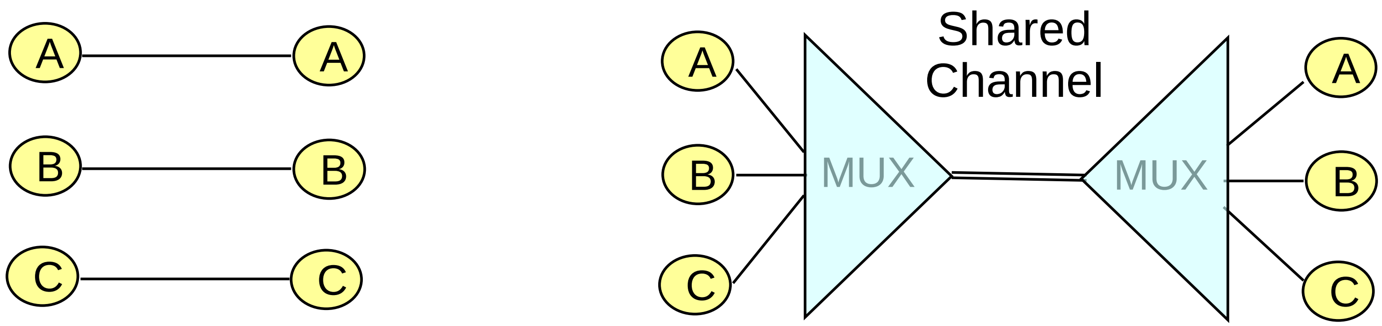
- After being sampled and quantized signal is coded, utilizing binary codebook, for transmission (the rest is the story about modulation and electromagnetic wave propagation)
- CODEC is a program/system that encodes and decodes digital data signal
- Audio codec refers to the device encoding an analog audio signal to a digital audio signal, or decoding an analog audio signal from a digital audio signal
- There are many variations of Audio/Voice codes as we will see in the upcoming lecture



Pulse Code Modulation (PCM)
DS0

Digital Signal Transmission – Multiplexing

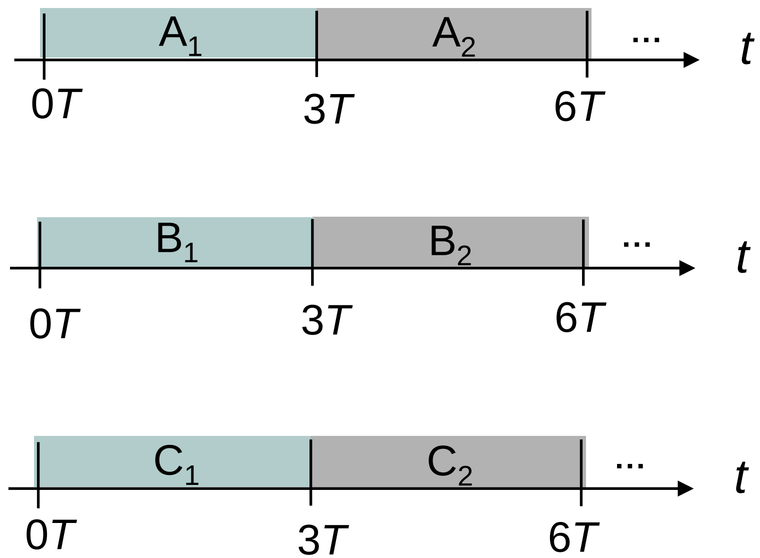
- Multiplexing involves the sharing of a transmission channel by several connections or information flows
  - Channel = 1 wire, 1 optical fiber, or 1 frequency band
- Significant economies of scale can be achieved by combining many signals into one
  - Fewer wires/pole; a fiber replaces thousands of cables



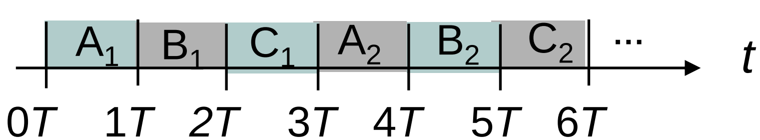
Time-Division Multiplexing - Telephone digital transmission

- High-speed digital channel divided into time slots

Each signal transmits 1 unit every 3T seconds

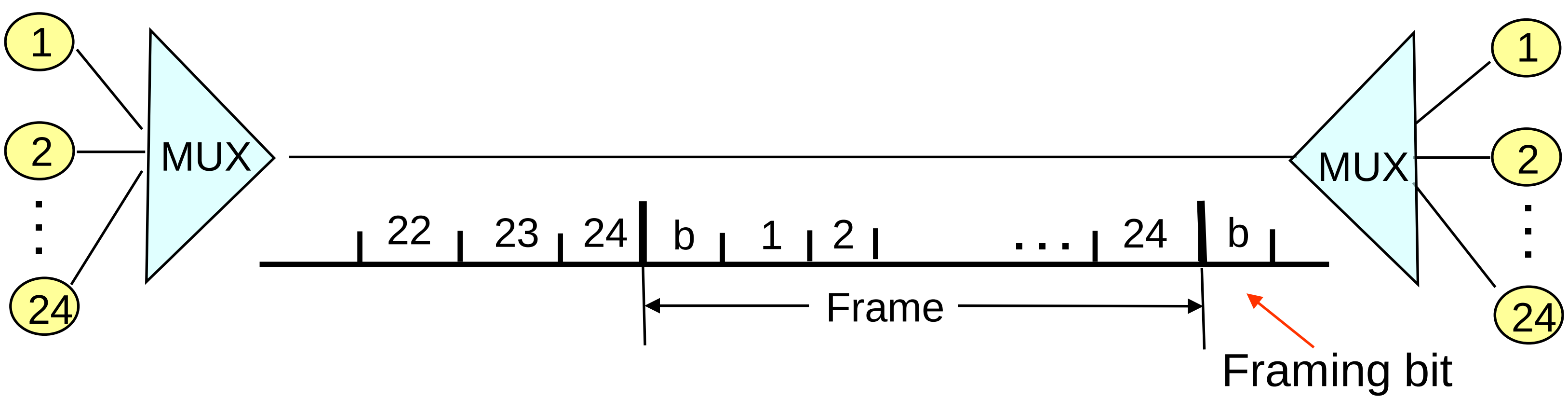


Combined signal transmits 1 unit every T seconds



T-Carrier System

- Digital telephone system uses TDM
- PCM voice channel is basic unit for TDM
  - 1 channel = 8 bits/sample x 8000 samples/sec = 64 kbps
- T-1 carrier carries Digital Signal 1 (DS-1) that combines 24 voice channels into a digital stream



Bit Rate = 8000 frames/sec. x (1 + 8 x 24) bits/frame = 1.544 Mbps

Voice Transmission

- PCM yields to the net composite bandwidth for transmission of 64,000 bits per second or 64kbs (8000 samples/second x 8 bits/sample). This is the standard transmission rate for one channel of telephone digital communications

- For a voice call, this level is generally considered to be "toll quality‟, or the quality of voice that one would anticipate receiving from a commercial telephone carrier

- Voice TDM circuit-based transport (adequate for a constant rate transmission):
  - T1/E1
  - DS3/E3
  - SONET (OC-3, OC-12, etc.)

- Voice packet-based transport:
  - ATM
  - Frame Relay
  - **IP** ← **Main subject for this course**

Before we continue - Terminology

- **PSTN:** Public Switch Telephone Network
- **POTS:** Plain Old Telephone Service
- **LEC:** Local Exchange Carrier
- **CLEC:** Competitive Local Exchange Carrier
- **ILEC:** Incumbent Local Exchange Carrier
- **LATA:** Local Access and Transport Area
- **IXC:** Inter-Exchange Carrier
- **PBX:** Private Branch eXchange
- **Bandwidth:** Line capacity (in KHz or **Kbits/s**)
- **Voice Circuit:** The bandwidth used by a voice communication
- **IN Services (or Class Services):** 800 Number, LNP (Local Number Portability), Call Forward, etc.

Legacy Telephony Architectures

- The Public Switched Telephony Network (PSTN) is a global circuit-switched network that was designed primarily for voice traffic (residential and inter-enterprise telephony)
- GSTN – General Switched Telephone Network as defined by ITU-T

- Enterprise Telephony is a business telephone system that provides basic business features (call-hold, three-way calling, etc.)

- In common:
  - Circuit switching based on TDM
  - Common Infrastructure Model (call control, bearer channel, phones connect directly into a switch)
  - Services they offer

- Difference:
  - # of local loops – pstn switch >100,000, pbx switch > 5,000
  - The way they treat signaling- pbx (proprietary) but it uses CAS and PRI as interface to pstn, pstn uses SS7, ISDN and in-band signaling links

As we will see during the semester VoIP term covers Carrier-grade IP telephony and Internet Telephony as a subset (special case) for enterprise and residential telephony operational scenarios.

PSTN Overview - History

- Graham Bell connected 2 rudimentary phones (Carbon Membrane, Battery and a Magnet) with an electrical cord in 1876 - this was a direct connection between 2 phones with no dialing

- Design evolved from one-way to a bi-directional voice transmission

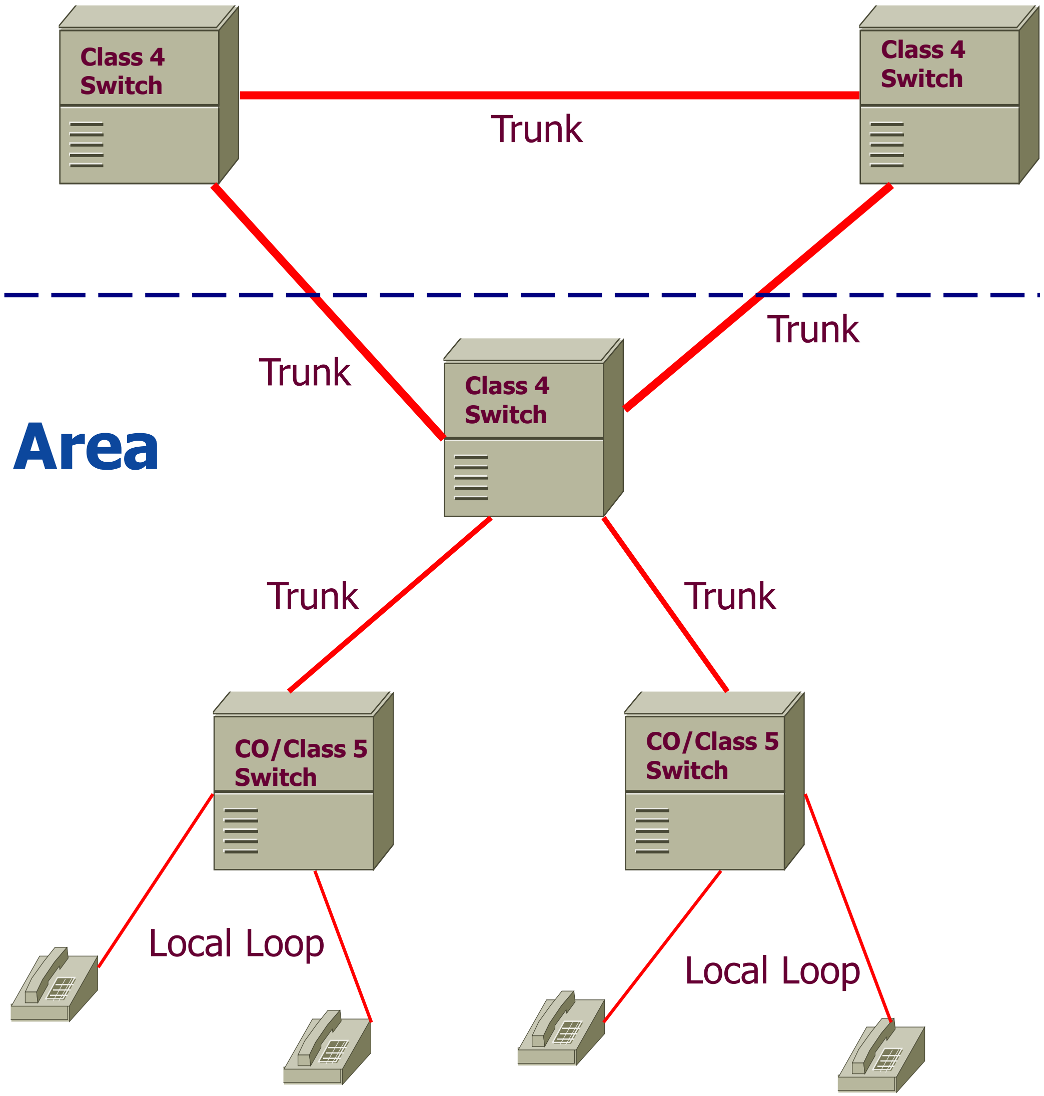
- First improvement: connect every phone to an operator to switch calls

- Second improvement: Dialing and Mechanical Switches (later on: Electronic Switches)

PSTN Overview – Architecture

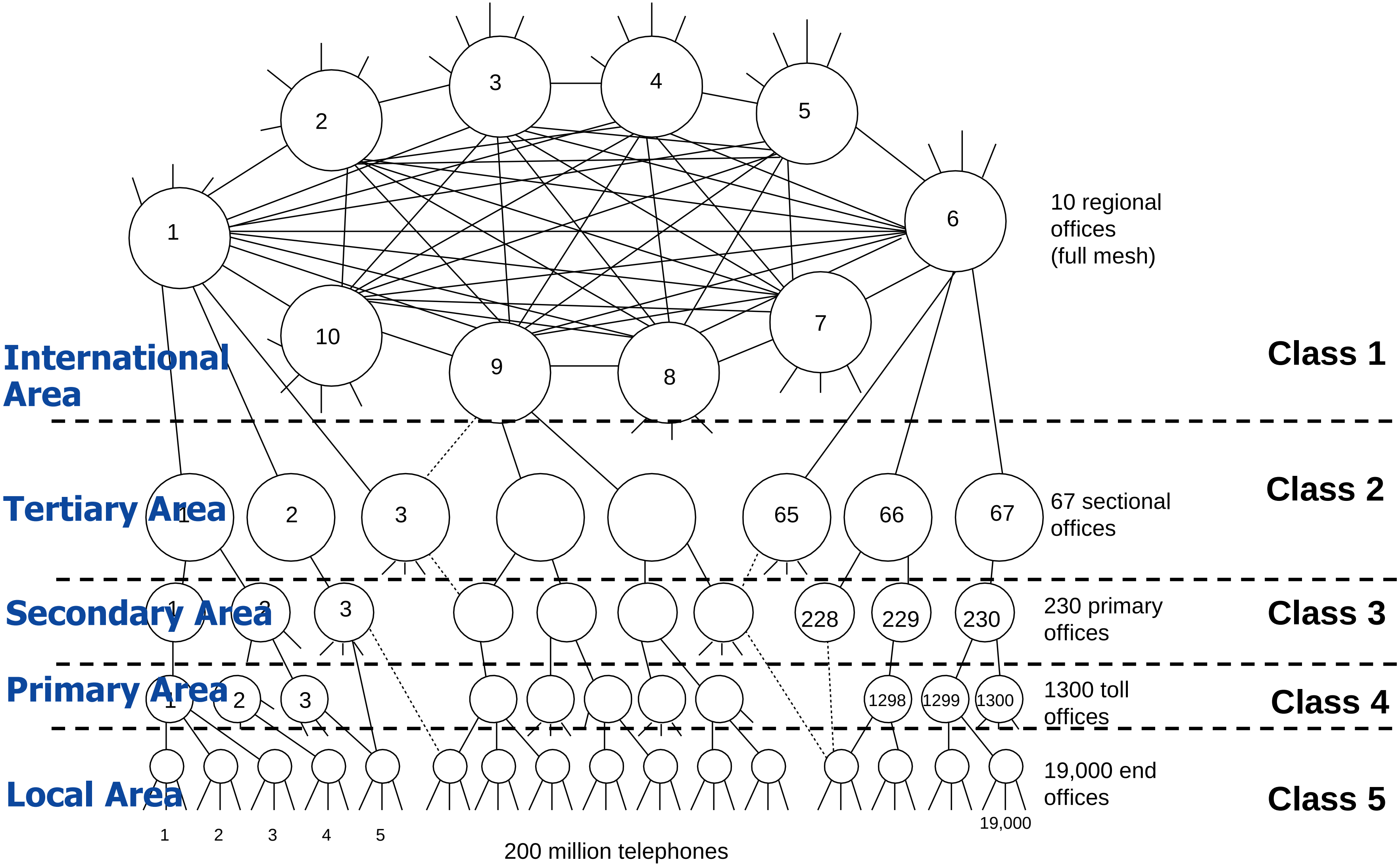
- A pair of copper wires (Local Loop) runs between a subscriber home and a local Central Office (CO or Class 5 switch)
  - Exchange – second name for a voice switch
- COs connect to their local Tandem Switch (or Class 4 Switch)
- Local Tandem Switches connect to higher Layer Tandem Switches
- Switches connect through Trunks
- Same portions of the PSTN use as many as five levels of hierarchy
- Traffic categories
  - Upstream or downstream
  - Incoming, outgoing, internal, terminating, originating, transit
- Service Nodes – connected to the edge of the telecom network
  - Voice mail systems
  - Voice response systems
  - Announcement devices
  - Service Control Points (SCPs)
  - Space for competition

PSTN Hierarchical Layout



- Trunk is the link between switches
  - Usually copper cables between local switches
  - Optical Fiber between higher level switches
  - Trunks carry digital voice
- Local Loop is the link between a subscriber and the Central Office
  - Voice stream is usually analog
  - The capacity (or bandwidth) of the line is limited to 4KHz (64KBits/s)
  - ISDN (digital voice, out of band signaling) did not really catch up

Typical PSTN Hierarchy



End-to-end connection can have max of 12 circuits – addressing is hierarchical

Source: Computer Networks, Andrew S. Tanenbaum

PSTN Service Model

- Permanent circuit is set up on demand
- Transfer capacity can be used as best or as poorly
- Customer pays based on used network resources
  - Usage-based billing
  - Time-based billing
  - Providers attach some other fees (sometimes explainable/sometimes not) ☺

Signaling Concepts

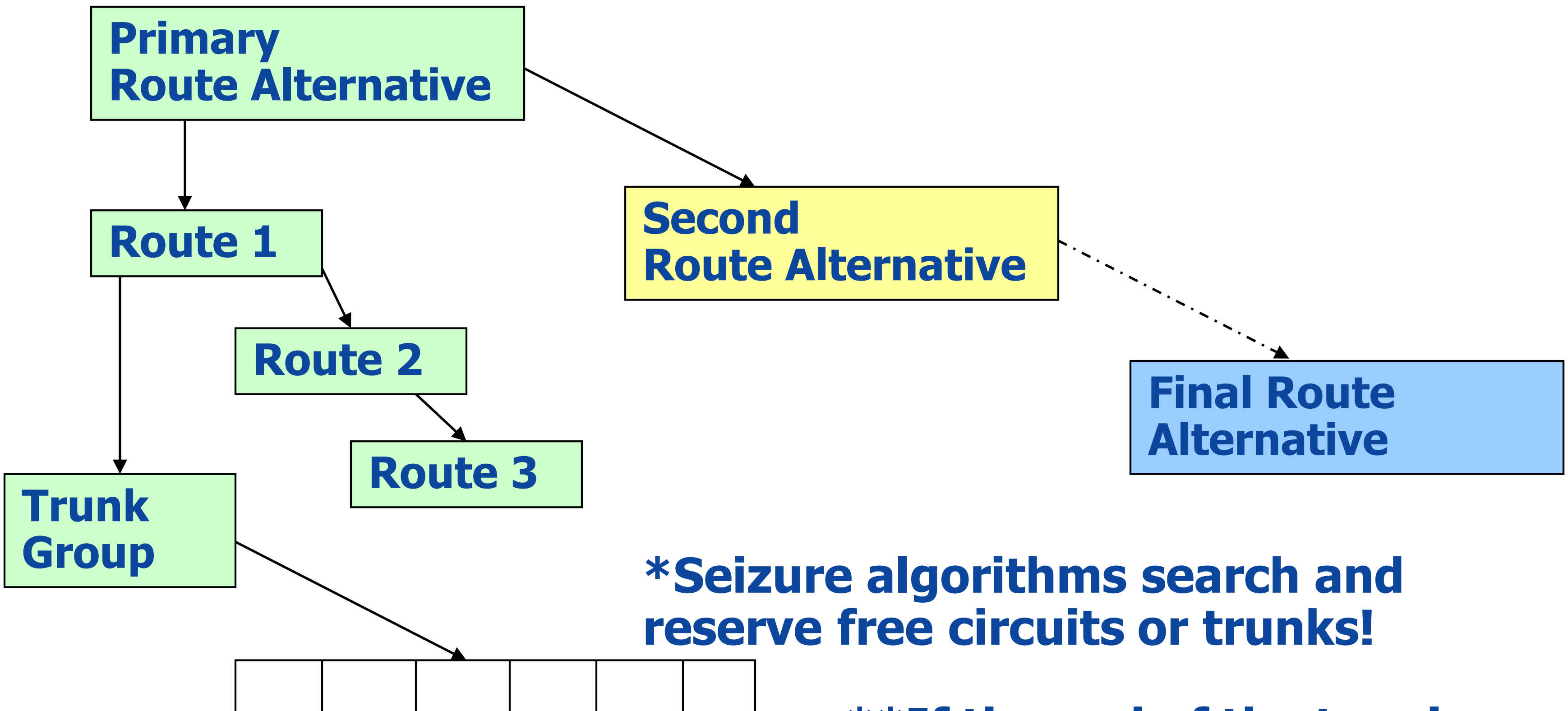
- Signaling is the generation, transmission, and reception of information needed to direct and control the setup and disconnect of a call
- Two groups of signaling methods
  - User-to-network signaling – end user/PSTN signaling – Pulse, Dual Tone Multi Frequency, ISDN (BRI and PRI)
  - Network-to-network signaling (trunk signaling) – intercommunication between PSTN switches
- Signaling: on hook, off hook, digits collection
- In-band Signaling
- Out of Band Signaling

PSTN Signaling

- Network-to-network/Signaling between trunks:
  - Old Switches: in-band is not flexible, hard to deploy IN services
  - Modern Switch use out of band Signaling. SS7 is the most popular
  - SS7:
    - Very flexible
    - Saves bandwidth for voice
    - Delegates IN Services to special nodes in the network called Signaling Control Points (SCPs)

Call Routing

- Number analysis links the information received from signaling to call routing (dialed digits, incoming circuit info)
  - Analysis returns a set of routing alternatives or an instruction to perform number translation (800, 866, etc. numbers)



***Seizure algorithms search and reserve free circuits or trunks!**

****If the end of the tree is reached, the call is blocked**

PSTN Players

- Service Providers:
  - Local Services: LECs (Local Exchange Carriers). Baby Bells
    IN services: Caller Id, Call Blocking, Call Waiting Call Back, Last calls, Call Forwarding
  - Long Distance: IXCs (Inter-Exchange Carriers). Sprint, AT&T Inc., MCI (before its absorption by Verizon), CenturyLink, etc.
    - In Europe and throughout the rest of the world, the same PTT operators also usually provide inter-exchange service within their country
  - IN Services: 800/888/866 Numbers, Calling Cards
- Equipment Vendors: Ericsson, Nortel, Alcatel, Lucent and others

PSTN Legacy - Carrier Grade

- When was the last time you couldn't get dial tone in your home (assuming you are **still** using the analog legacy PSTN telephony system)
- Occasional "backhoe fade", but it almost always works
- Compare that to your cable service, or network access at work
- Over 100 years of engineering goes into the phone network, and things like 911 have become a critical service
- Data networking is much less mature
  - Computing has barely been around since ~1950
  - Serious data networking? ~1960s

PSTN Legacy - Carrier Grade (2)

- HI RELIABILITY
  - 99.999% - "5 Nines"
  - Becoming a requirement on data networking vendors who want to sell equipment to the telco world ☺
- SCALABILITY
  - Support 10s/100s of millions of end-users
  - Support 100s of thousands of simultaneous calls
  - Note the dependence on statistics
    - we expect that not everyone will want to talk on the phone at the same time
    - we also expect that the calls will be relatively short
    - Data modems started to violate these assumptions

PSTN Legacy - Carrier Grade (3)

- EFFECTIVENESS
  - To quote James Earl Jones, “The network’s no good unless the call goes through”
  - You dial and you get through in less then 1 or 2 seconds
  - Conversation is perceptible (MOS scores – later)
- Lots of standards
- High interoperability
- Support structure in place

Enterprise Telephony

- Private companies have different needs than residential users: more services, short dialing
  - With VoIP this gap has been closed

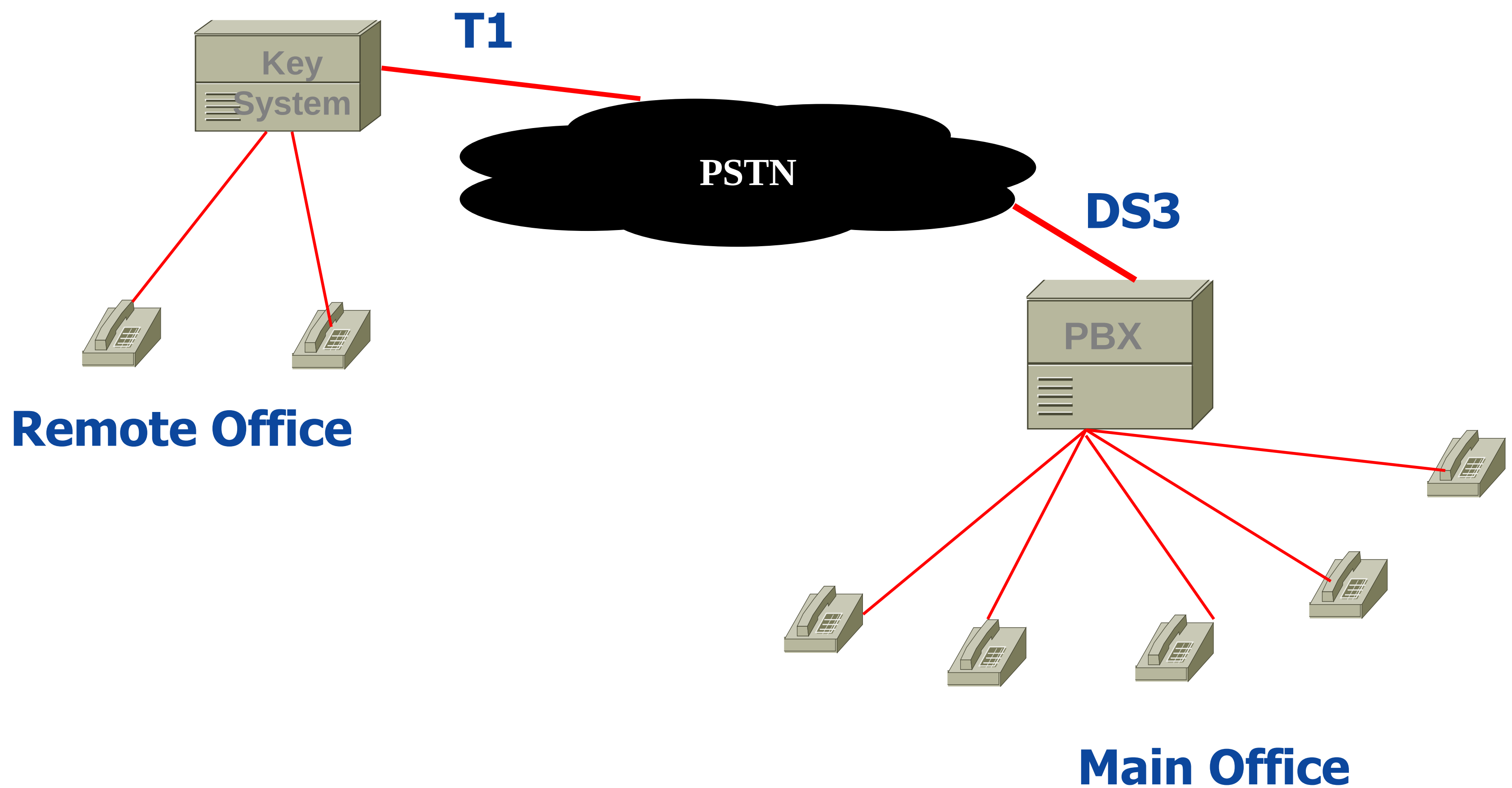
- Options:
  - Small Business Lines (SBL): higher monthly fees, limited capabilities
  - Centrex Lines: Local PSTN Provider manages Telephony Exchange (TE). Costly and not flexible but offers more features (transfer calls, calls on hold)
  - VPNs: Private network where telephone company manages a private dialing plan
  - Acquire own Switch or PBX. Flexibility to add, move, numbers. Key Systems are small PBXs

Enterprise Telephony – PBX

- Business circuit switched telephony system with business features
  - Call Hold
  - Three way calling
  - Call transfer
  - Forwarding

- PBX usually provides a programming interface: CTI (Computer Telephony Integration) to support additional applications: Call Centers, Conferencing, etc.

Enterprise Telephony - PBX Example

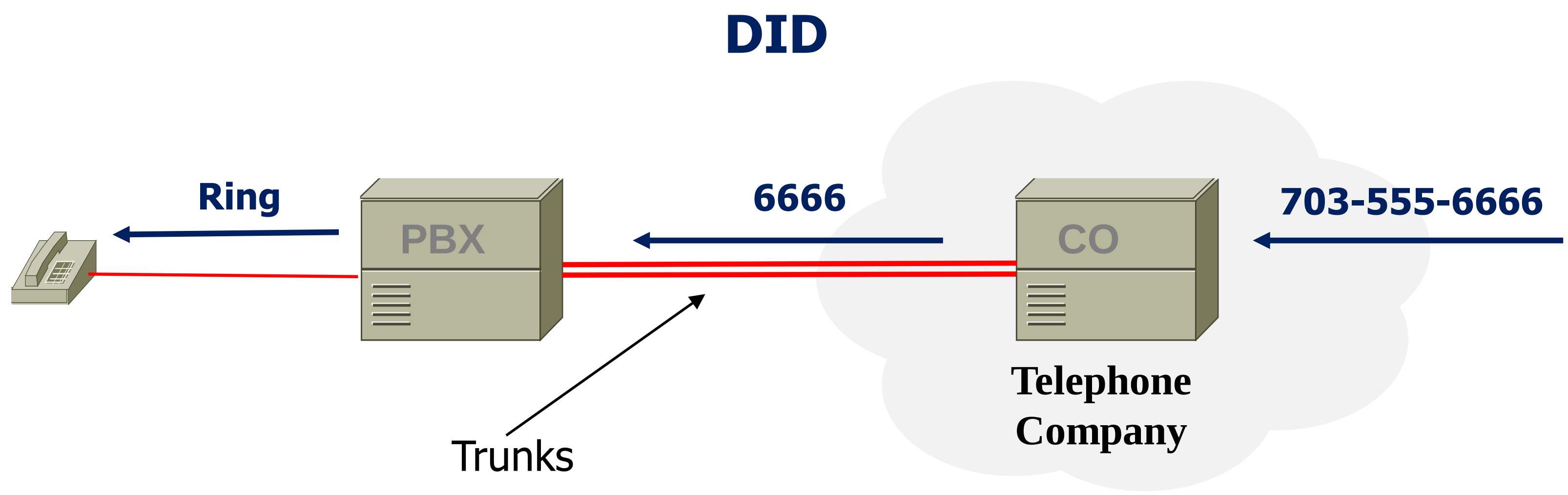


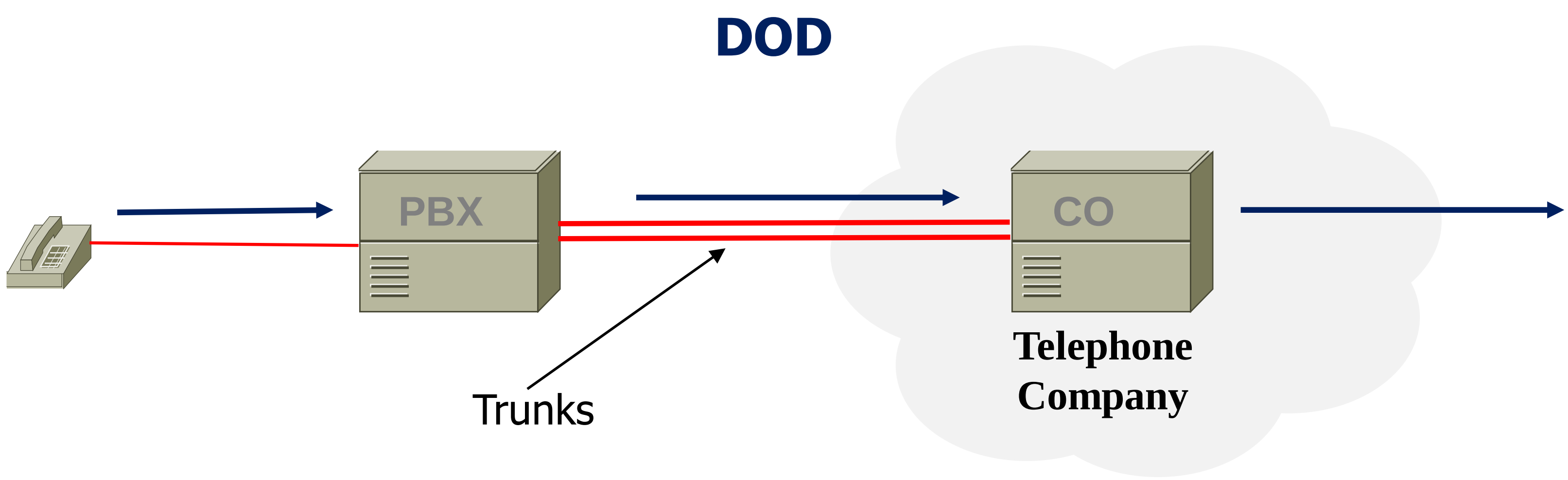
Direct Inward Dial (DID)

- In DID service the telephone company (LEC) provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk
  - As calls are placed to the PBX, the dialed destination number, Dialed Number Identification Service (DNIS) is transmitted, usually partially (e.g., last four digits), so that the PBX can route the call directly to the desired telephone extension within the organization without the need for an operator or attendant
  - The service allows direct inward call routing to each extension while maintaining only a limited number of subscriber lines to satisfy the average concurrent usage of the customer

- DID service is usually combined with direct outward dialing (DOD) allowing PBX extensions direct outbound calling capability with identification of their DID number
  - For example, if you want to place a call outside your company's network, you simply dial an access code such as 9, and the PBX forwards your call out to the CO. At that time, the CO provides a second dial tone and uses the remaining dialed digits to forward the call to the final destination.

DID/DOD Diagram





Pros and Cons of Voice Telephony

- Pros:
  - Reliable and Excellent quality
  - Built in support mechanisms
  - Secure and robust enough for DoD
- Cons:
  - Steep learning curve
  - Expansive to manage and upgrade
  - Cumbersome for phone Add/Move/Change

Telephony Regulations and Brief History
(not covered during the class: slides 60 - 66)

Telecommunications Regulations

- Federal Communications Commission (FCC) regulations cover telephony, cable, broadcast TV, wireless etc. in United States
  - http://www.fcc.gov/
- "Common Carrier": provider offers conduit for a fee and does not control the content
  - Customer controls content/destination of transmission & assumes criminal/civil responsibility for content
- Local monopolies formed by AT&T's acquisition of independent telephone companies in early 20th century
  - Regulation forced because they were deemed natural monopolies (only one player possible in market due to enormous sunk cost)
  - FCC regulates interstate calls and state commissions regulate intra-state and local calls
  - Bells + 1000 independents interconnected & expanded

Telephony Industry - Beginnings

- In 1876 Graham Bell formed Bell Telephone which licensed local telephone exchanges in major US cities
- AT&T was formed in 1885 to connect the local Bell companies
- In 1912 AT&T agreed to become regulated monopoly. In exchange they had to connect competing local companies and let Federal Communication Commission (FCC) approve their prices and policies

Deregulation of Telephony

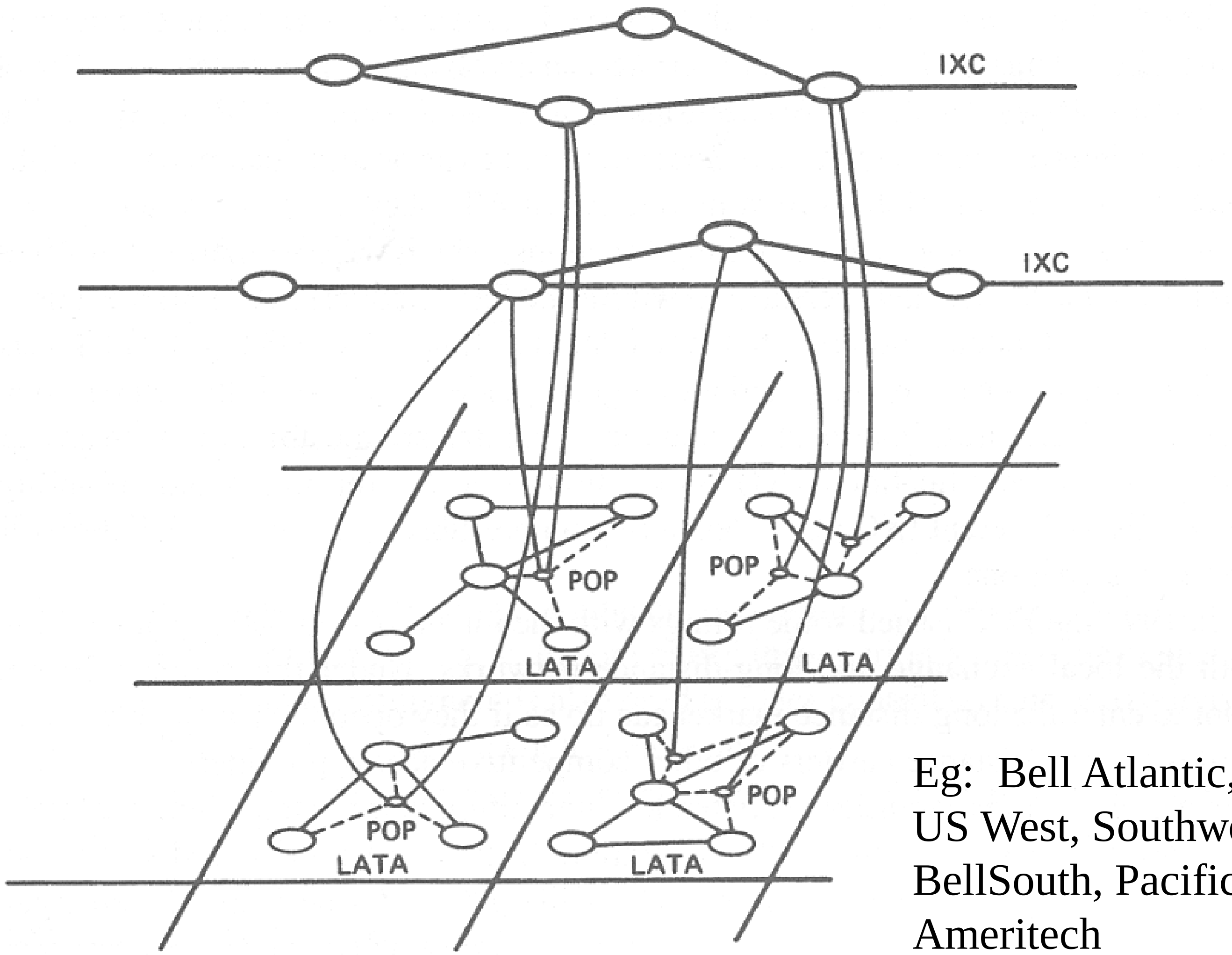
- 1960s-80s: gradual de-regulation of AT&T due to technological advances
  - Terminal equipment could be owned by customers (CPE) => explosion in PBXs, fax machines, handsets
  - Modified Final Judgement (MFJ): breakup of AT&T into ILECs (incumbent local exchange carrier) and IXC (inter-exchange carrier) part
    - Long-distance opened to competition, only the local part regulated…
    - Equal access for IXCs to the ILEC network
    - 1+ long-distance number introduced then…
  - 800-number portability: switching IXCs => retain 800 number

- On January 1st, 1984, a court forced AT&T to give up its 22 local Bell companies, establishing seven Regional Bell Operating Companies (RBOC)

- 1995: removed price controls on AT&T

US Telephone Network Structure (after 1984)

Eg: AT&T, Sprint, MCI



Eg: Bell Atlantic, Nynex, US West, Southwestern Bell, BellSouth, Pacific Telesys, Ameritech

Telecom Act of 1996

- Required ILECs to open their markets through unbundling of network elements (UNE-P), facilities ownership of CLECs
  - Today UNE-P is one of the most profitable for AT&T and other long-distance players in the local market: due to apparently below-cost regulated prices
- ILECs could compete in long-distance after demonstrating opening of markets
  - Only now some ILECs are aggressively entering long distance markets
  - CLECs failed due to a variety of reason
- ILECs still retain over 90% of local market
- Wireless substitution has caused ILECs to develop wireless business units
- VoIP driven cable telephony + wireless telephony => more demand elasticity for local services

VoIP Era

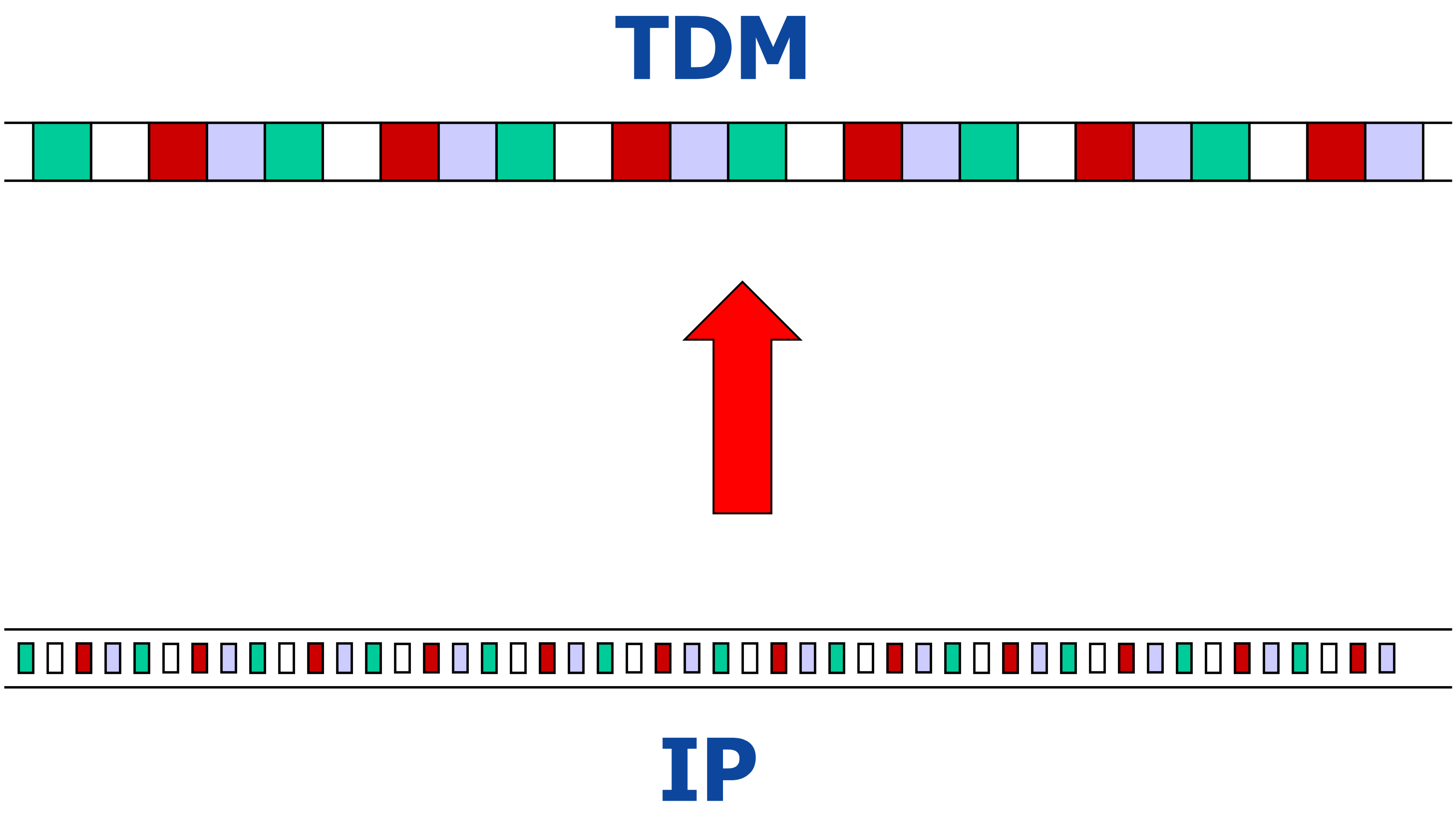
- In February 2004, the FCC ruled that electric power companies could use their wiring for Internet service, including voice over IP (VoIP). They also ruled that companies providing computer-to-computer VoIP service should not be subject to the same regulations as telephone companies. (This ruling did not apply to companies that operate gateways between the Internet and the telephone network)

- The RBOCs remain dominant, although they are under threat from wireless operators, cable companies, VoIP providers, etc.

Rest of the World

- Other countries have also introduced private ownership and competition in the telephone industry
  - However telephone company was usually a government owned and operated monopoly
- Only two percent of world telecommunication revenue is generated by companies that are fully owned by the government
- In spite of privatization and the introduction of competition, the initial incumbent telephone companies remain very powerful, accounting for 85% of telephone company revenue

VoIP Switch (Lets talk about VoIP)



What does VoIP mean?

- Voice using IP
  - but not necessarily over the Internet
- We'll see that VoIP requires a great deal of **control** over the network
  - especially to assure quality of service
- Once you go out over the global Internet:
  - variety of link speeds
  - variety of carriers and policies
  - best-effort service
  - increased complexity

What does VoIP mean? (2)

- The control is not existent in the Interne but outside the Internet efforts to provide control exist and are continuously evolving – we'll look at some of them (use of QoS and MPLS and other network control mechanisms)

So, Why VoIP?

- "If it not's broken, don't fix it"
  - some truth to this statement
  - telephony is not "broken"
  - so what is "broken"?
- Why carry voice using IP networks? First that comes to mind…
  - Internet has provided new services, especially integration of voice and data creates new service capabilities (Skype, WhatsApp, FaceTime, Uber, and many others)
  - There are potential $$$ in carrying voice

Why VoIP? (2)

- Why use IP for voice?
  - Circuit switching worked very well
  - But now we have at least two networks
    - The voice network – equipment that costs A LOT
    - The data network – the Internet – equipment that might not cost as much
  - If I could carry both types of service over the same network, wouldn’t that be a good thing?
    - The phone company already had this idea: B-ISDN – Broadband Integrated Services Digital Network
    - Operation and management savings

Why VoIP? (3)

- Lower Equipment Cost
  - Phone systems are generally proprietary
    - You buy a PBX and the handsets (phone) that go with it
    - If you buy an e-mail server program, does everyone have to use the same e-mail client??? (Outlook can work with Thunderbird, Gmail, Pine, Elm, Eudora, Netscape, etc.)
    - PCs running Linux can often be used unlike PSTN
  - Hard to develop third-party software for phone systems
    - The WWW is the complete opposite of this paradigm
  - Traditional telephony is like mainframes
  - IP networks
    - More open standards
    - More competition (for the most part)
    - Moore's Law tends to motivate data networking

Why VoIP? (4)

- Voice/video/data integration and advanced services
  - User interfaces with HTML (point-and-click) are much easier to deal with than keypad sequences (e.g., #33#7)
  - Unified Messaging
  - Easier new service introduction
- Potentially lower bandwidth requirements
  - New voice communications gear can take advantage of low-rate vocoders
  - Legacy telephone network largely stuck with 64 kbps per call
    - Some bandwidth reduction in some areas
  - Bandwidth management is better (for example: during silence periods there is no need for information transmission)

Why VoIP? (5)

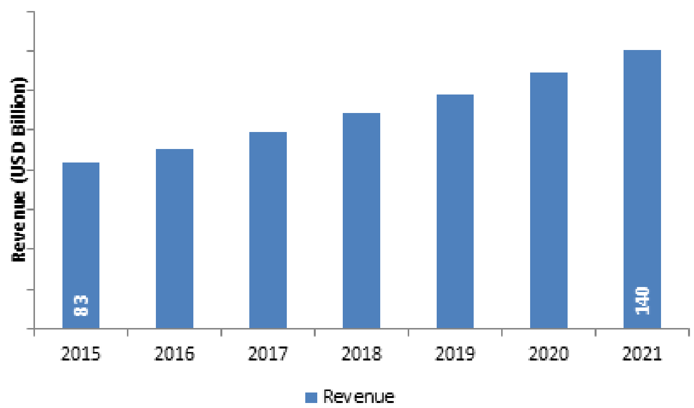
- Mobility
  - IP address and end-user not tied to a particular area or Service Provide
- Widespread availability of IP
  - It's EVERYWHERE
    - But so are telephones ☺
  - Other packet-based mechanisms: FR, ATM
    - Not as present
- Circuit-switch networking product development has stopped
  - All R&D effort in telephony goes to VoIP telephony

Still to remember

- IP
  - has no guarantees
  - engineered for data traffic
    - data integrity is very important; TCP does a great job at this
    - latency – and VARIABLE latency – was not a major concern in the design and engineering of TCP/IP
    - asynchronous – can start, stop, tolerate variability in transmission speeds
- Quality of VoIP is generally lower than the legacy digital telephony but the gap is primarily closed
- Emergency calling features
  - PSTN has been better in handling 911 and power outages

VoIP Market

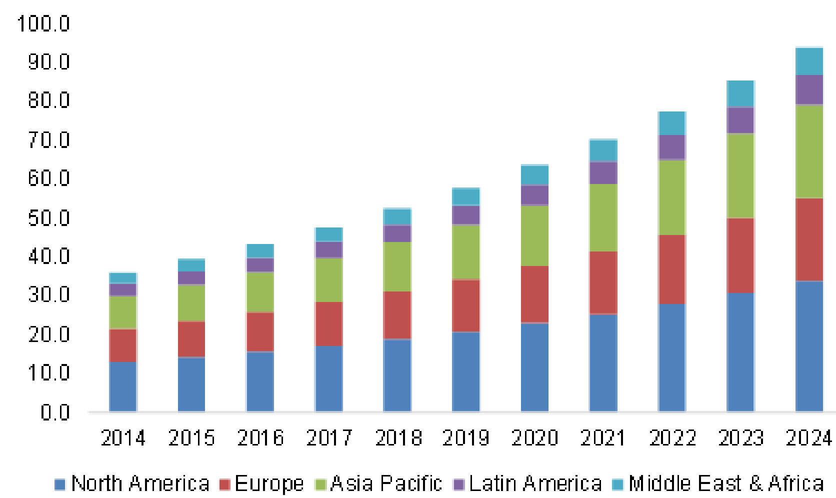
Global VoIP Market, 2015 – 2021 (USD Billion)



Source: Zion Research Analysis 2016

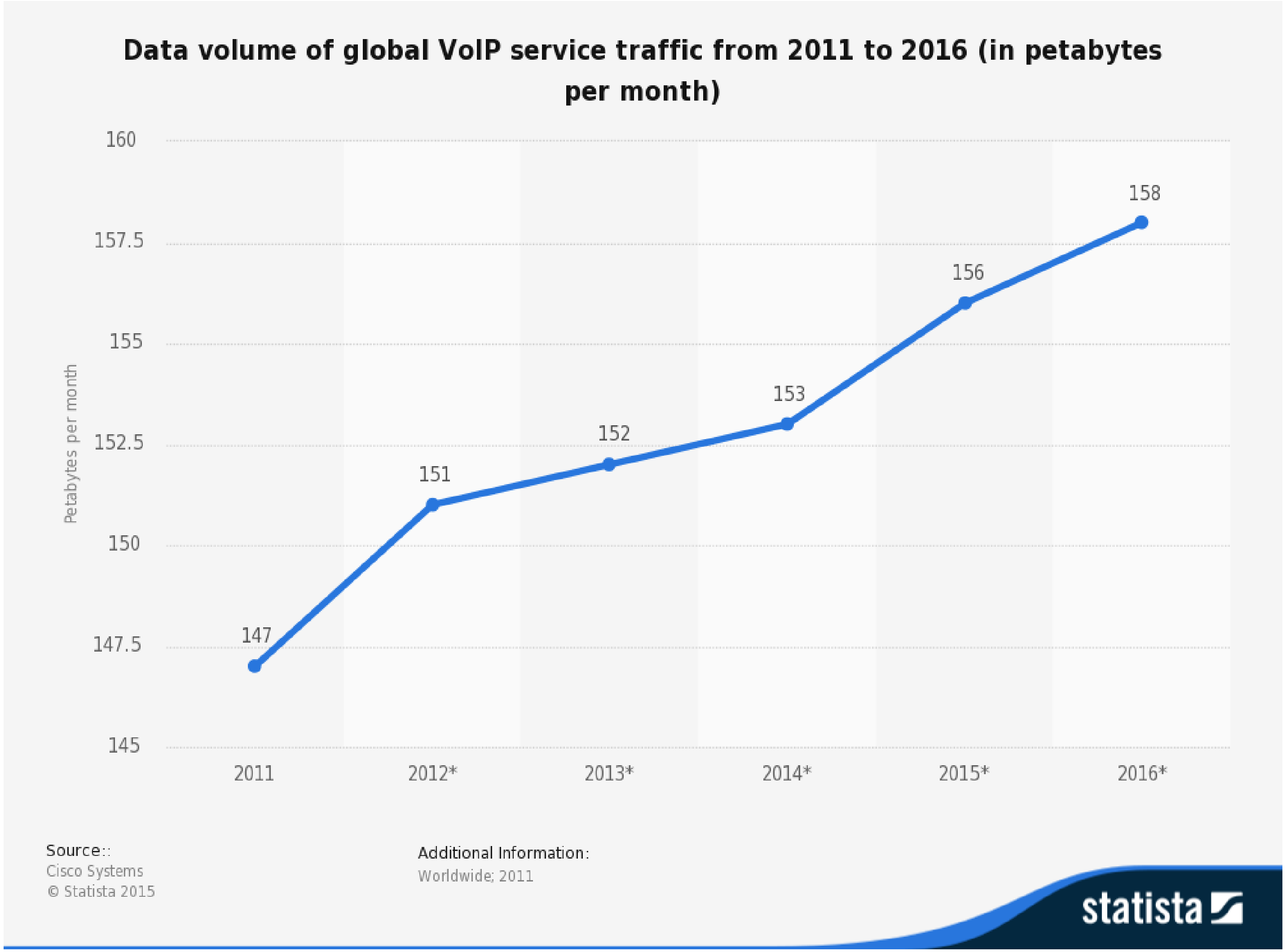
- VoIP technology has recently matured enough to deliver high-class services to residential and business users for the lower price.

Global Voice over Internet Protocol Market, by Region, 2014-2024 (in BN USD)



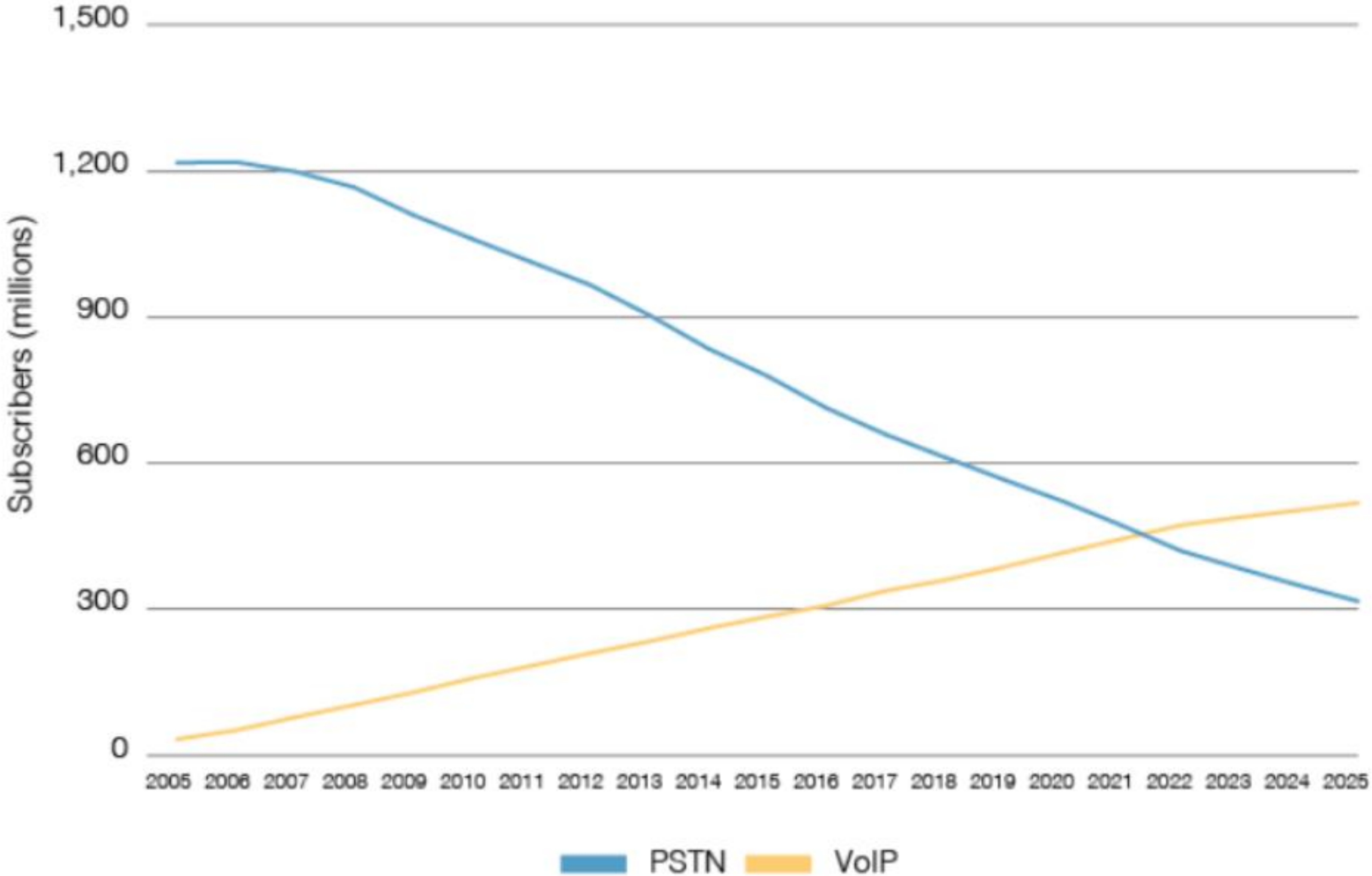
Source: Ameri Research

VoIP Demand



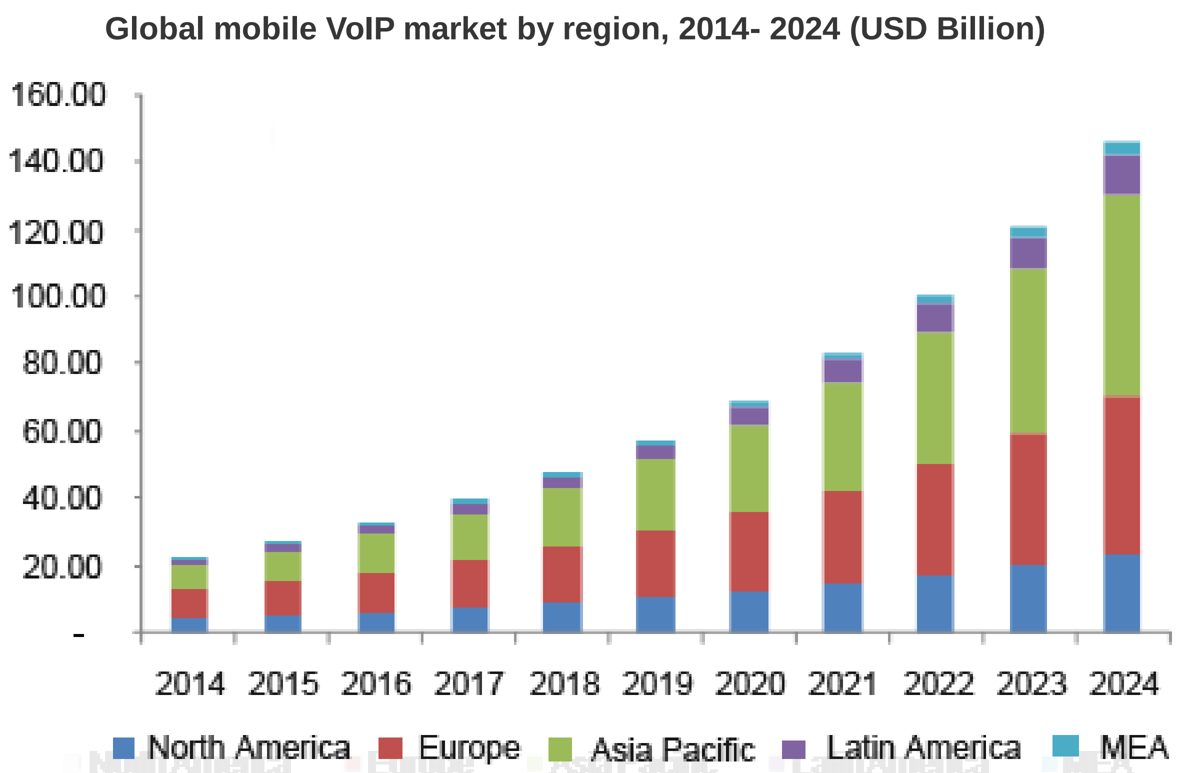
Data volume of global VoIP service traffic from 2011 to 2016 (in petabytes per month)
160
157.5
155
152.5
150
147.5
145
Petabytes per month
147
151
152
153
156
158
2011
2012*
2013*
2014*
2015*
2016*
Source::
Cisco Systems
© Statista 2015
Additional Information:
Worldwide; 2011
statista

Legacy vs. VoIP Traffic Trends



Source: telegeography.com

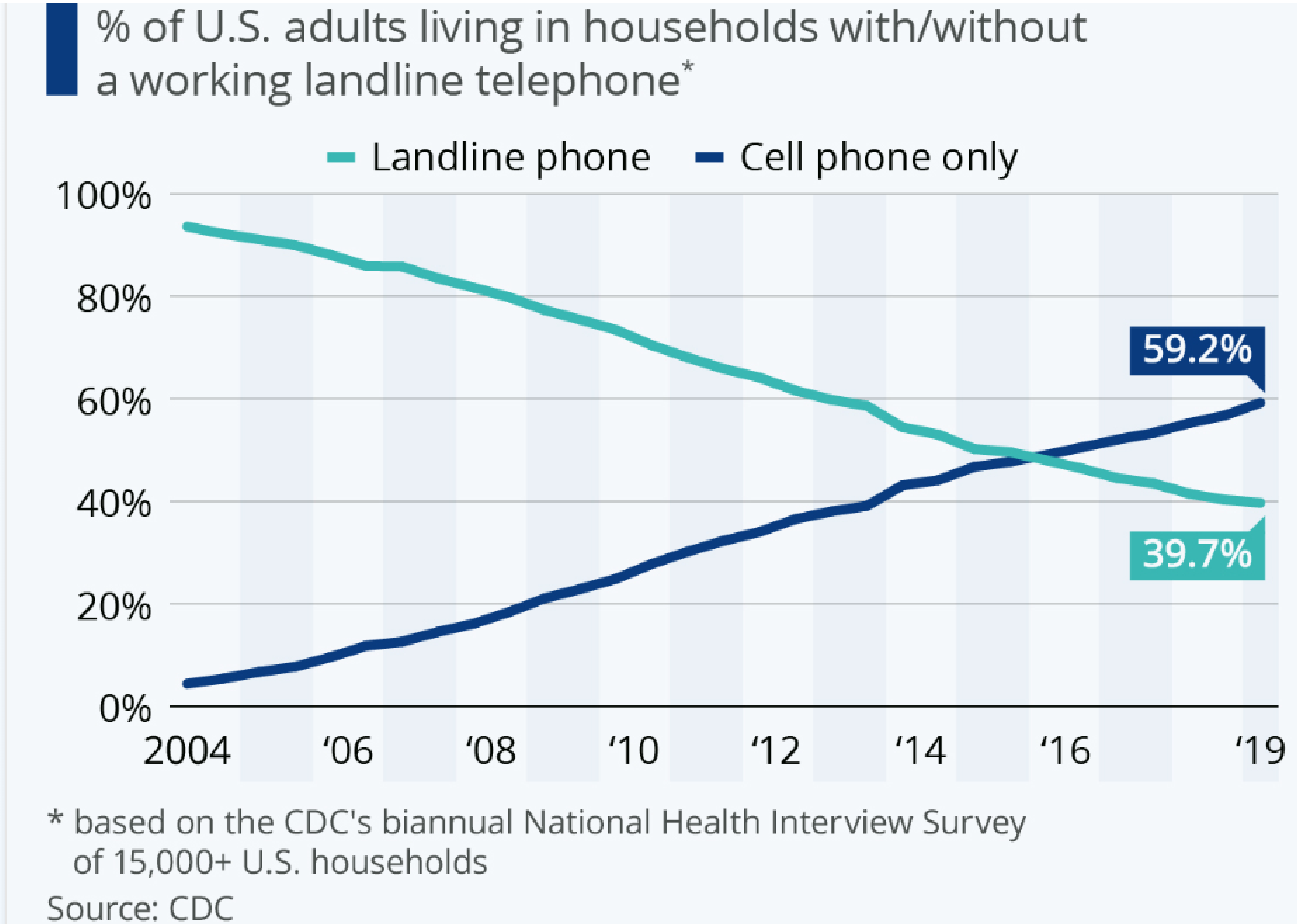
Growth of Mobile VoIP



Source: Grand View Research

The increase in penetration of smartphones has fueled the growth of mobile VoIP market. Smartphone manufacturers are developing devices that are compatible with VoIP services. The rising popularity of innovative and user-friendly applications including Skype, Viber, Line, Kick, WhatsApp and others are driving the demand for mobile VoIP services. Inexpensive data rates and the existence of robust network infrastructure will increase the growth of mobile VoIP.

Another Trend



Changing Expectations?

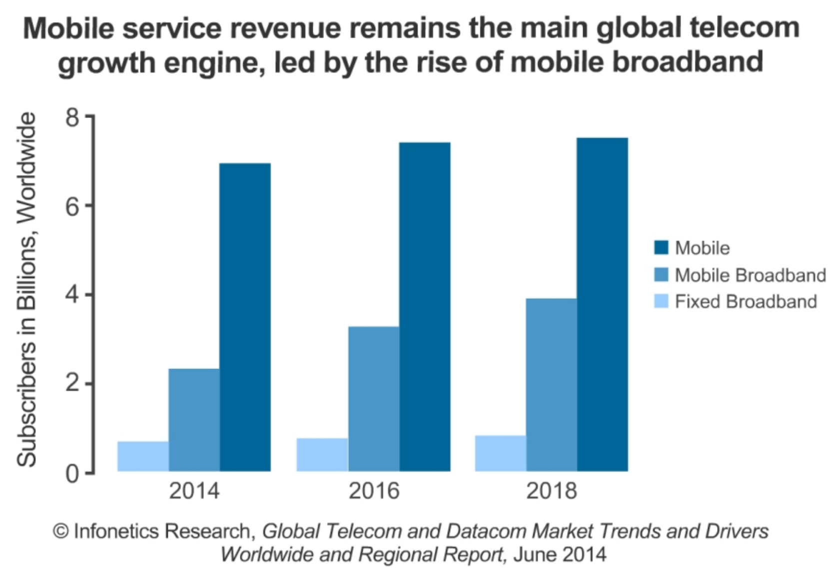
- THOUGHT 1: What has cellular telephony done to change our expectations regarding voice quality?
  - Tradeoff: mobility vs. availability
- THOUGHT 2: The rise of cloud computing - hosted voice systems eliminate the need to manage and maintain VoIP infrastructure, which makes IP-based telephony even more accessible

Introduction of VoIP – stretched road

- Takes cash flow from existing PSTN business models
- Time/usage based billing harder to maintain but it is not completely eliminated
- Cellular networks expansion – investing in wireline is no longer attractive
- User perspective on QoS and security of VoIP

Future Roadmap

- Data traffic grows over 30% a year and it is larger in volume
  - Mobile data traffic outnumbered legacy voice traffic
  - Broadband networks are packet-based technologies (ATM, IP…)

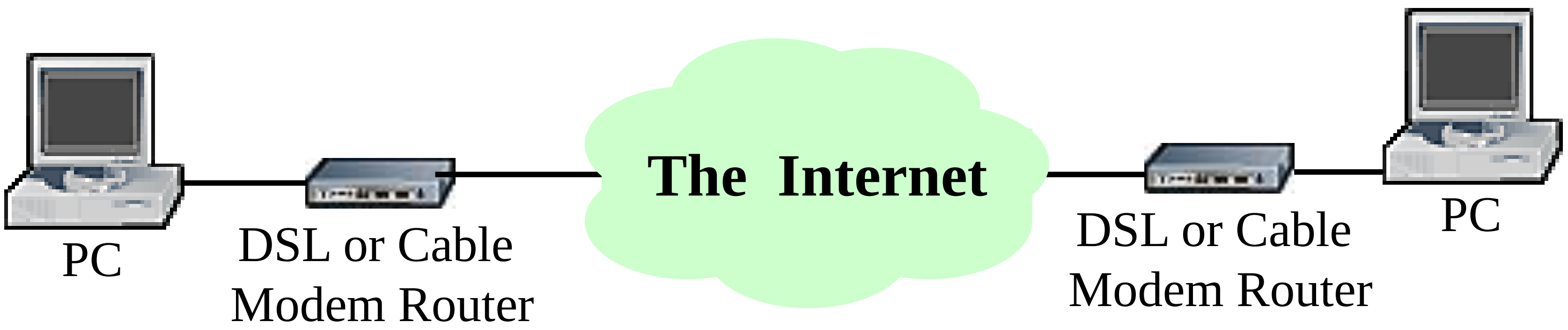


- Transport core network is moving to the packet technology
  - Ethernet, Metro Ethernet at 1GE, 10GE, 40GE, 100GE
  - IP/MPLS over OTN/WDM

Lets now introduce typical high-level VoIP architectures and connection strategies to get familiarized.

VoIP Solutions: Peer-to-Peer

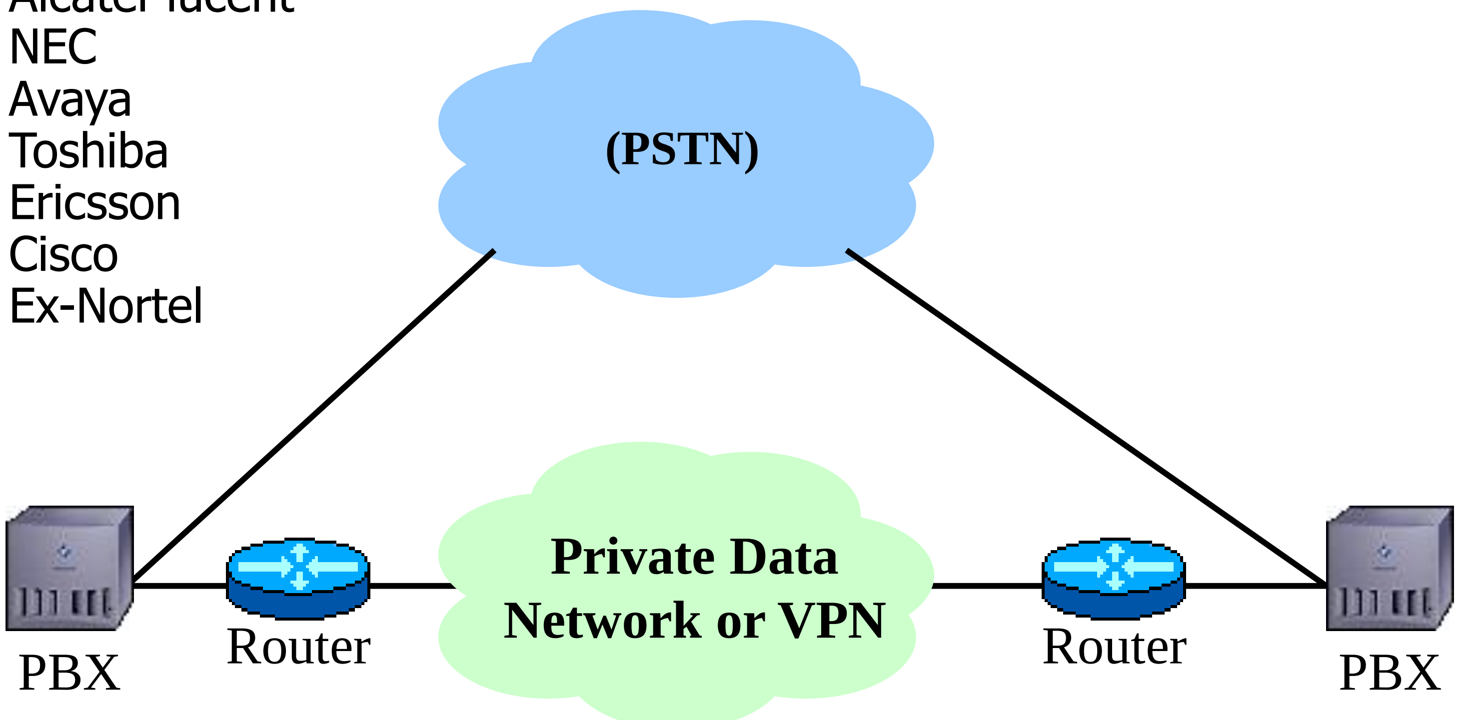
- Uses PC software to make calls over public and private internets
- It is free but no quality of service guarantees
- Some Earlier Players (not completely peer-to-peer any more):
  - Google Talk
  - Yahoo Messenger
  - Skype
  - Fring
  - Viber



VoIP Solutions: Enterprise – Toll Bypass

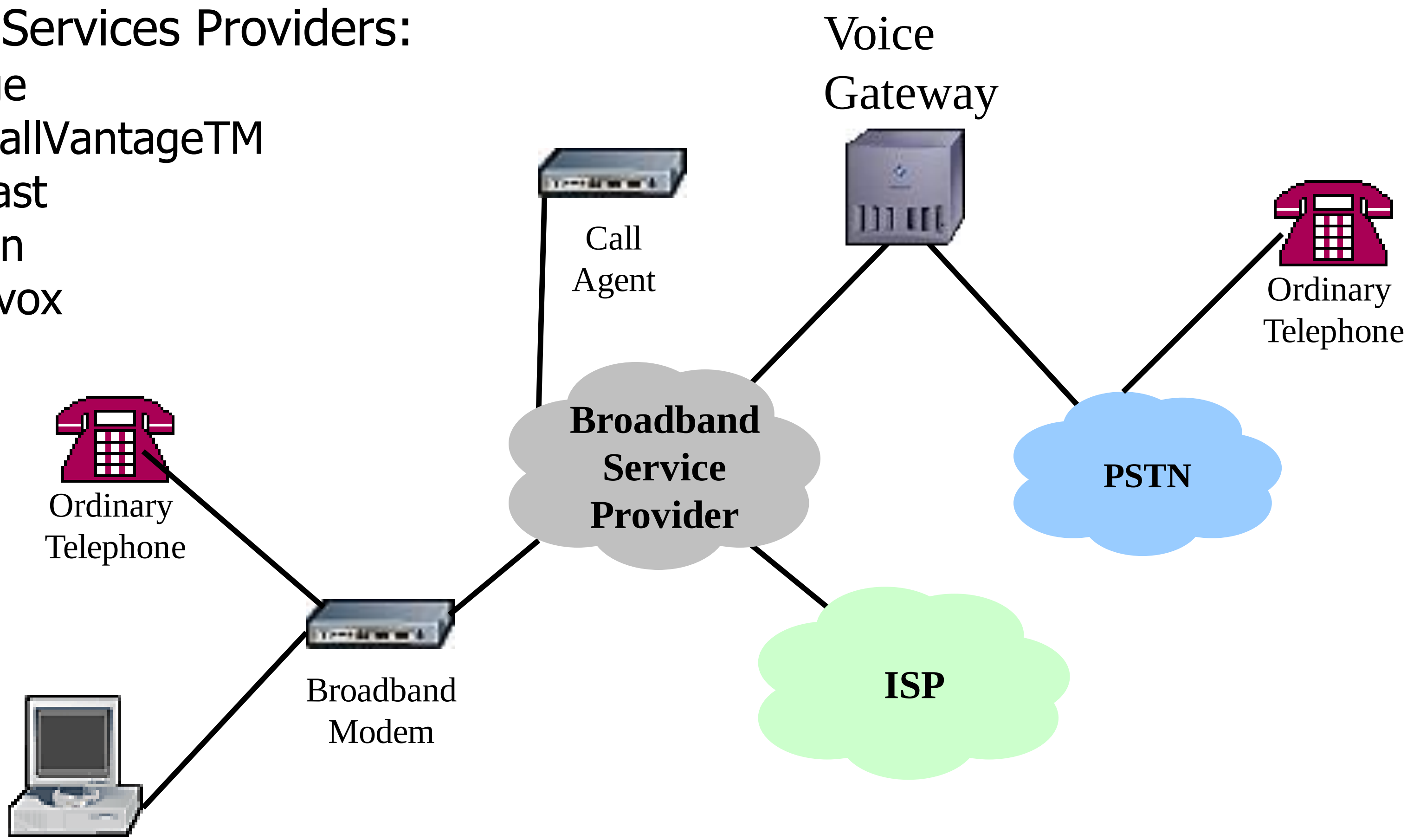
- Connecting enterprise PBXs with VoIP links to avoid paying for long distance charges

- Vendors:
  - Alcatel-lucent
  - NEC
  - Avaya
  - Toshiba
  - Ericsson
  - Cisco
  - Ex-Nortel

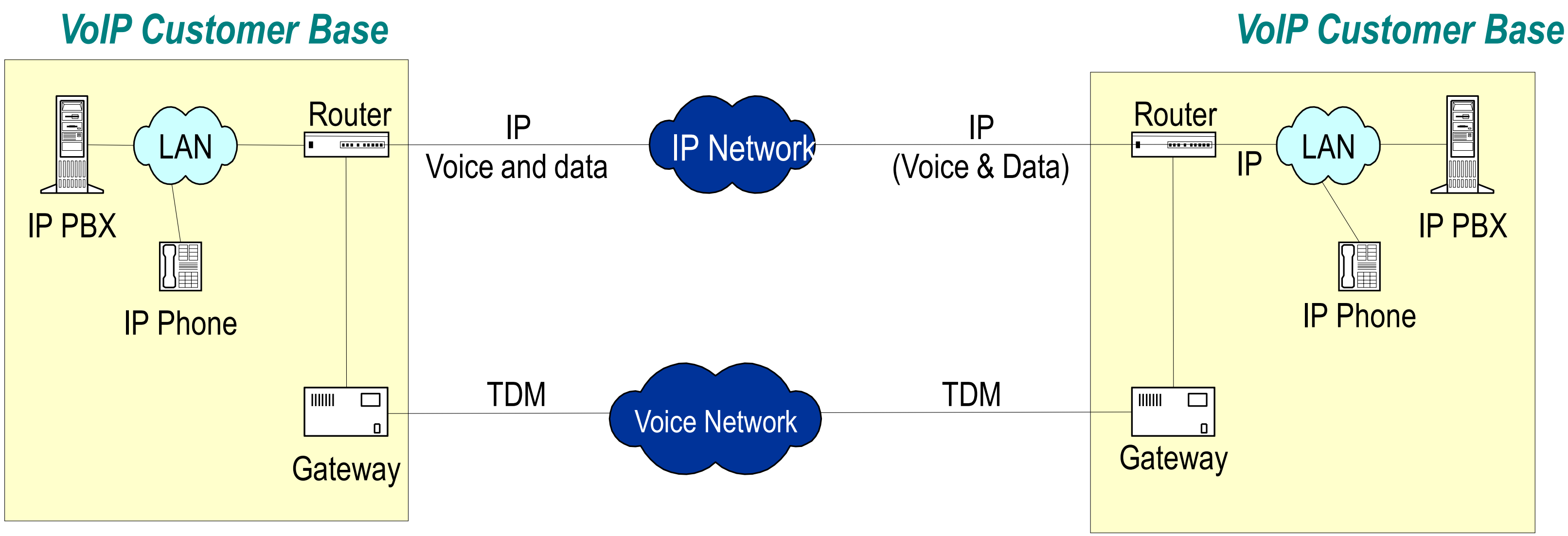


VoIP Solutions: Service Provider – Local Access

- Using broadband access to provide local and long distance telephone service
- Example Services Providers:
  - Vonage
  - ATT CallVantageTM
  - Comcast
  - Verizon
  - Broadvox
  - Cox

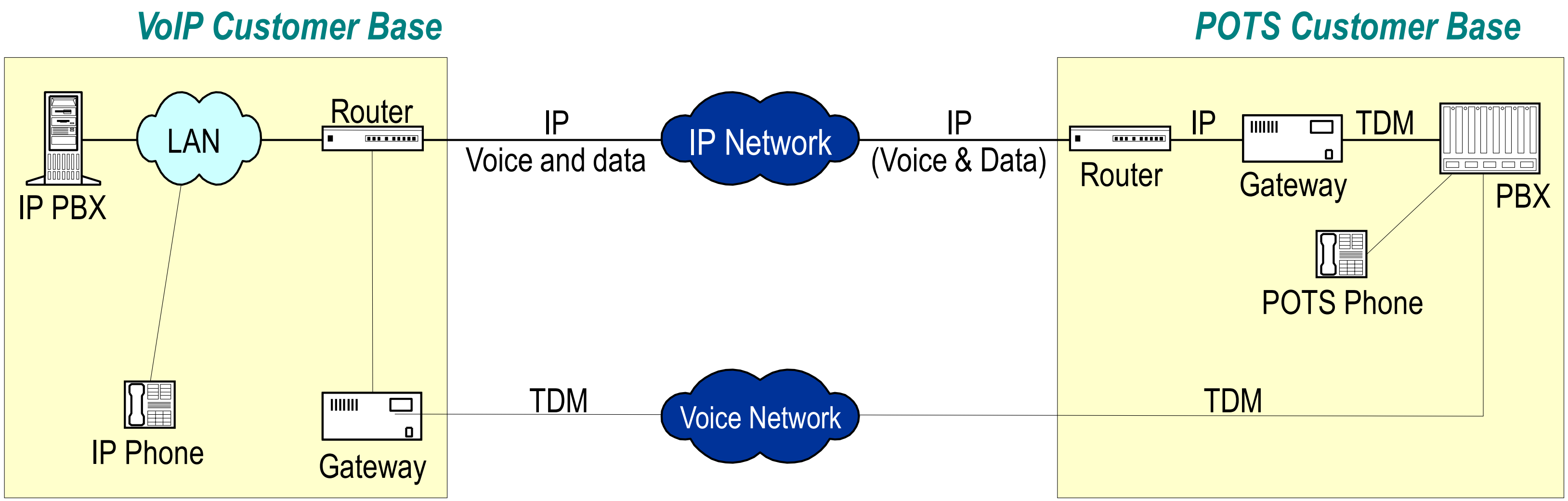


VoIP Connection Strategies



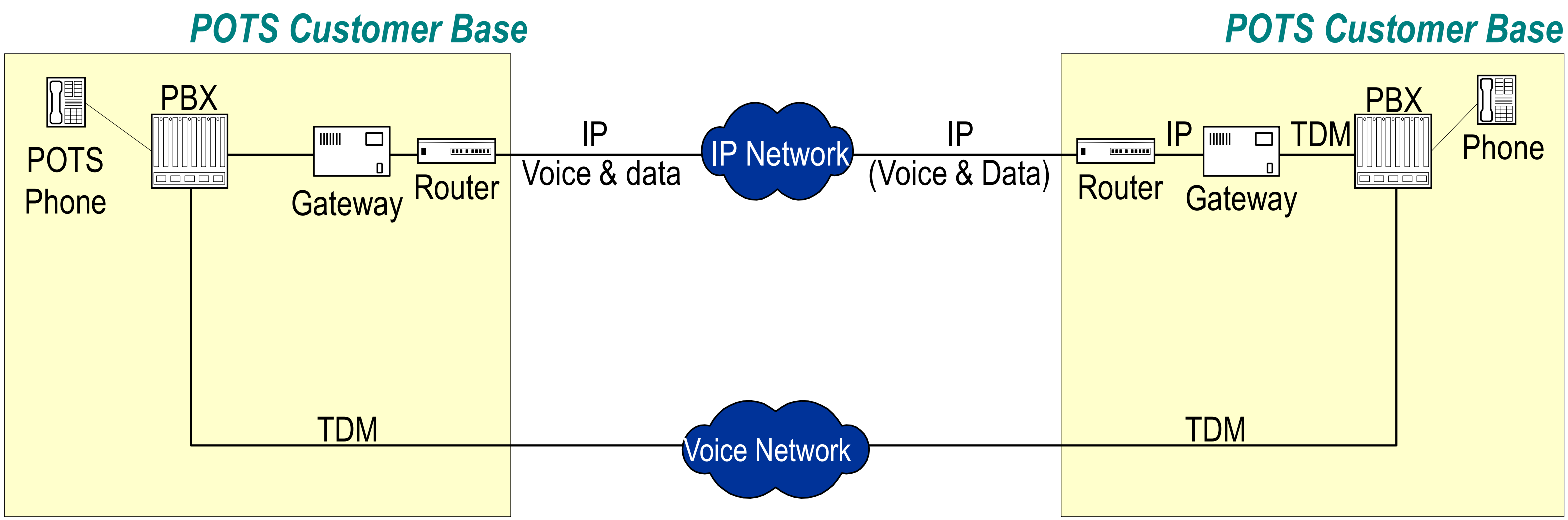
- Strategy 1: VoIP Base to VoIP Base over IP Network
- Strategy 2: VoIP Base to VoIP Base over Voice Network
- Next two slides present 4 other basic scenarios
  - Hybrids of the six basic scenarios are possible

VoIP Connection Strategies (cont.)



- Strategy 3: VoIP Base to POTS Base over IP Network
- Strategy 4: VoIP Base to POTS Base over Voice Network

VoIP Connection Strategies (cont.)



- Strategy 5: POTS Base to POTS Base over IP Network
- Strategy 6: POTS Base to POTS Base over PSTN