Voice over IP Tutorial

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Introduction

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What is VoIP?

- Legacy Telephony
  - TDM/SS7 based infrastructure
  - Traditional Class 5/Class 4 switches
- Voice over IP
  - IP-based packet infrastructure for PSTN voice transport
  - New elements that collectively perform traditional functions and more
- And what is IP Telephony?
  - Voice + Messaging + Video + Data over IP networks = IP Telephony
  - Public Internet : Best Effort Service
  - Managed IP Network : SLA based Service

The Packetization Process

Analog Voice (400-4000kHz) → Filter lower/higher frequencies → Sample at 8000 samples per second → Quantization → Digital Encoding into bytes

→ Compression (Optional). Yields further bandwidth gains → Silence Suppression (Optional). Yields ~50% bandwidth gains → Echo Cancellation(if reqd.)

→ Digitized Voice Packet encapsulated within RTP packet → RTP packet encapsulated within UDP packet → UDP packet encapsulated within IP packet

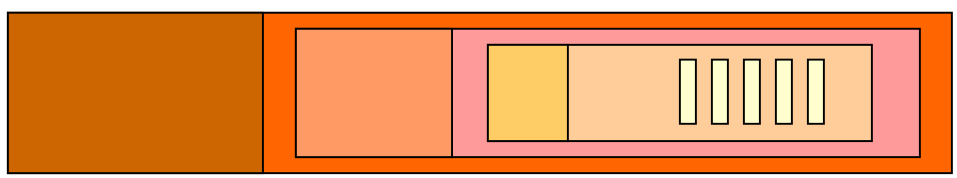
Voice Samples

RTP Payload — RTP Header

UDP Payload — UDP Header

IP Payload — IP Header

Voice Over IP Packet



Traditional PSTN Network

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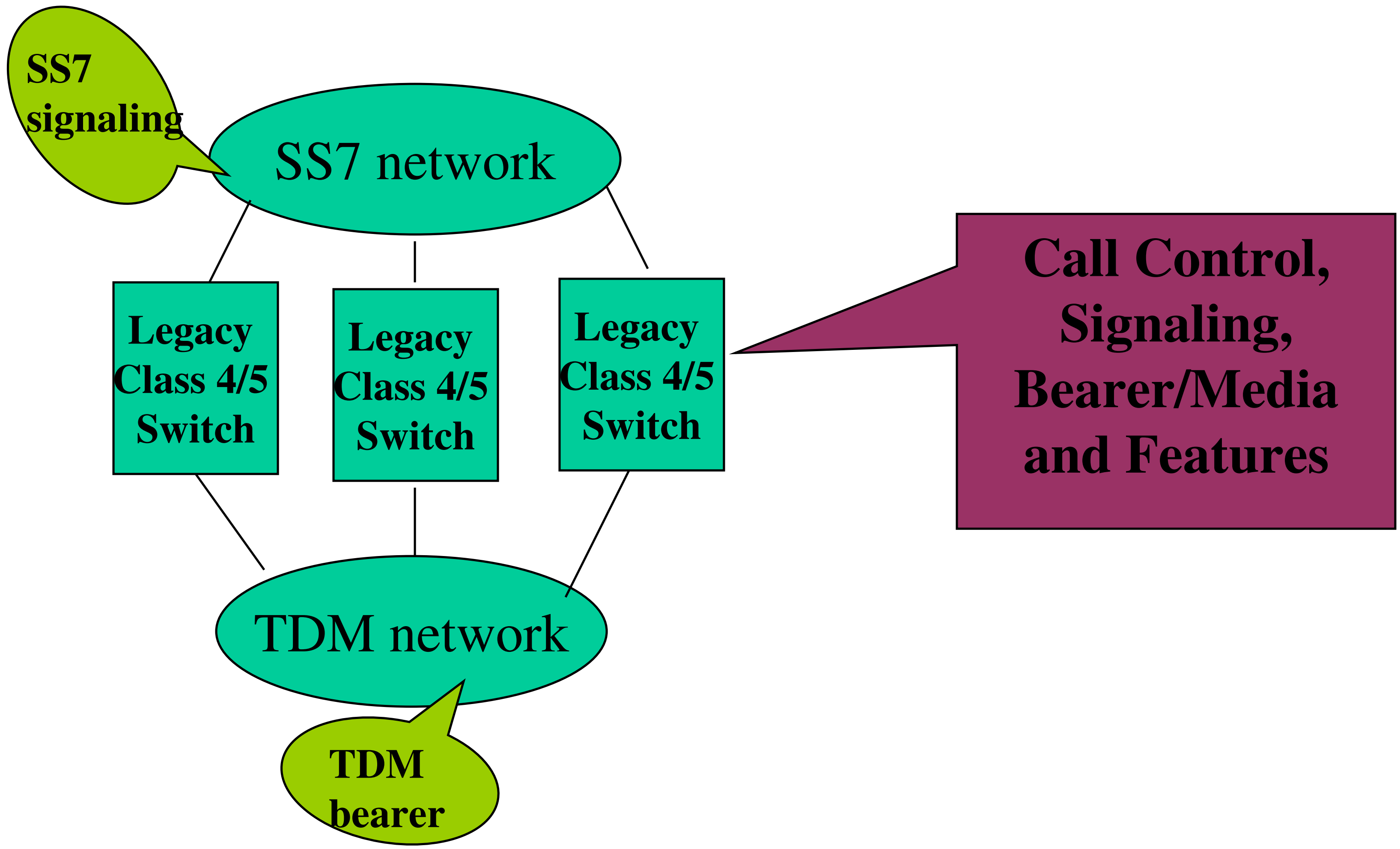
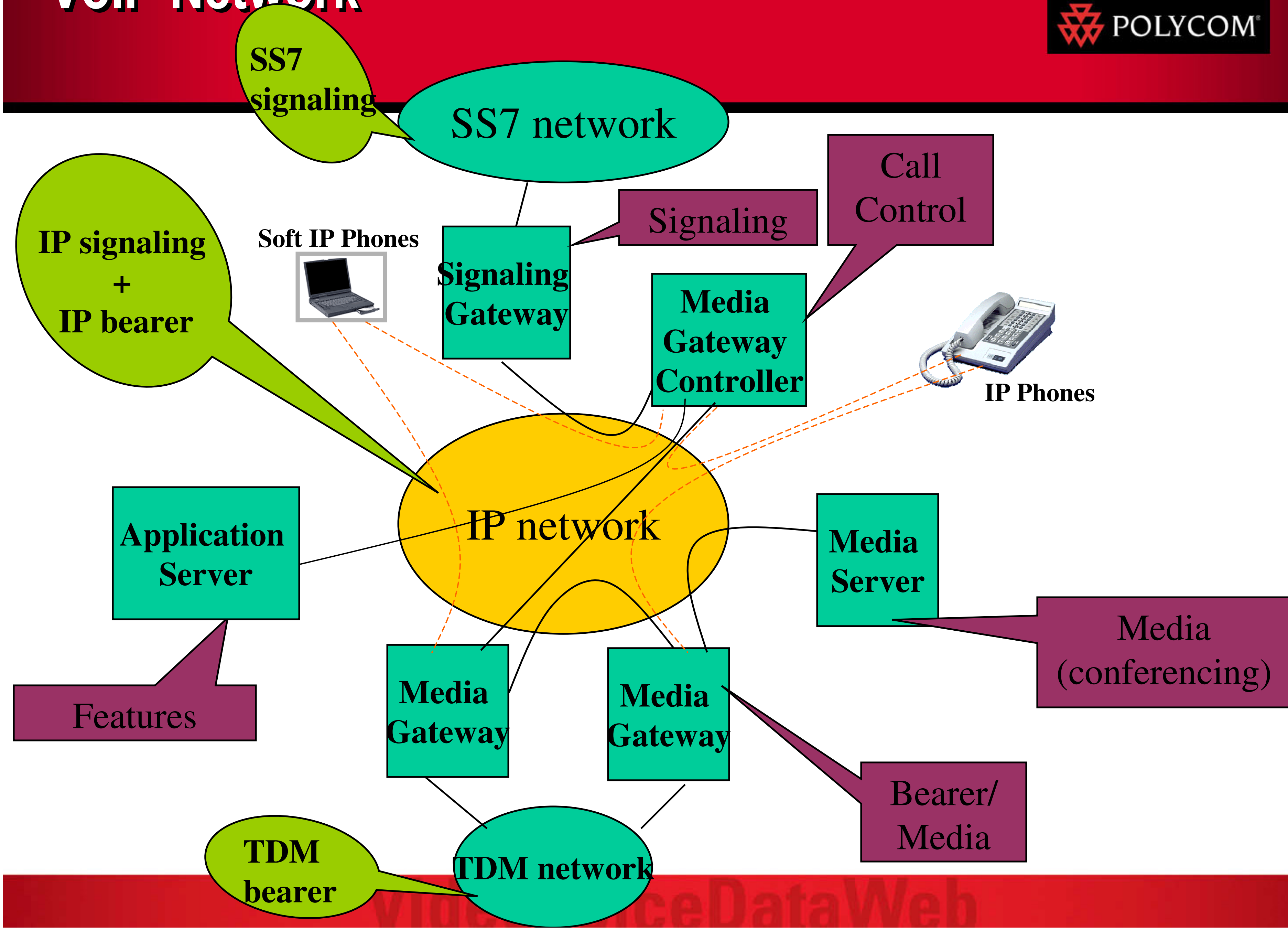


VoIP Network



VoIP Network Components

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The Elements ..

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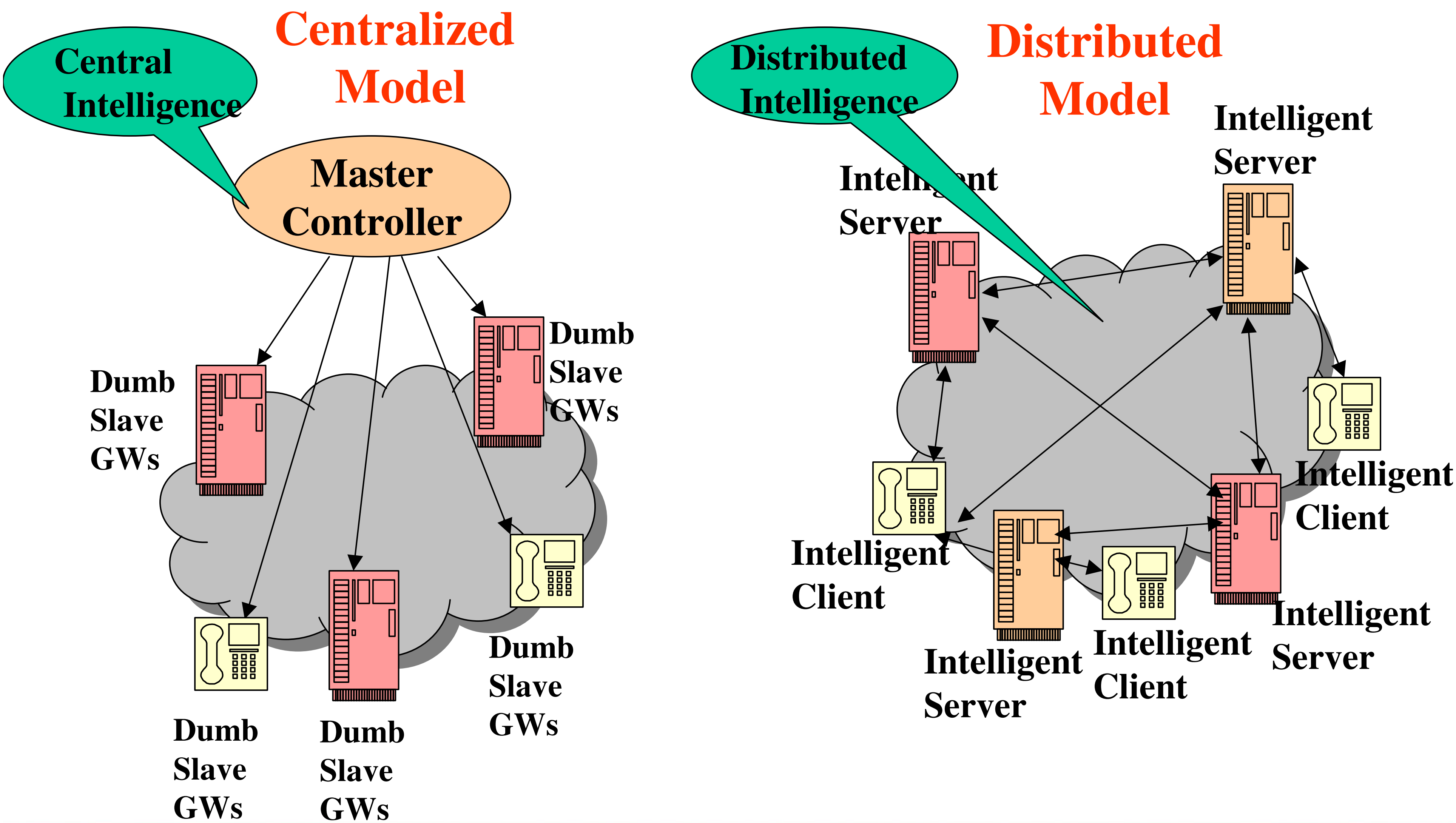
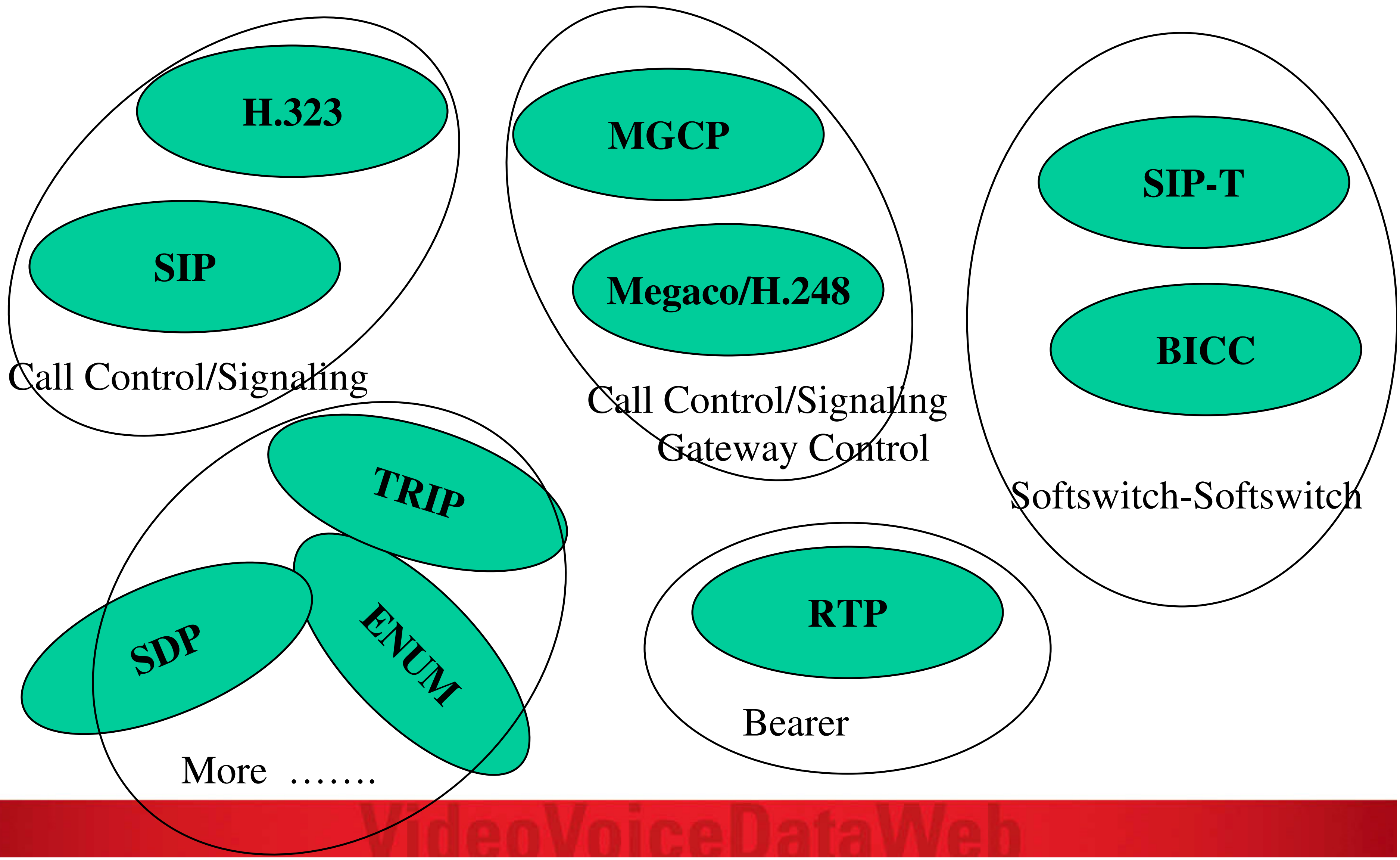
- Terminals or Endpoints
  - IP Phones
  - Soft Phones/PC Phones
- Media converter
  - Media Gateway/PSTN Gateway
- Call Processor
  - Media Gateway Controller or Gatekeeper or Proxy Server or Softswitch
- Signaling Gateway
- Application Server
- Media Server

VoIP Network Paradigms



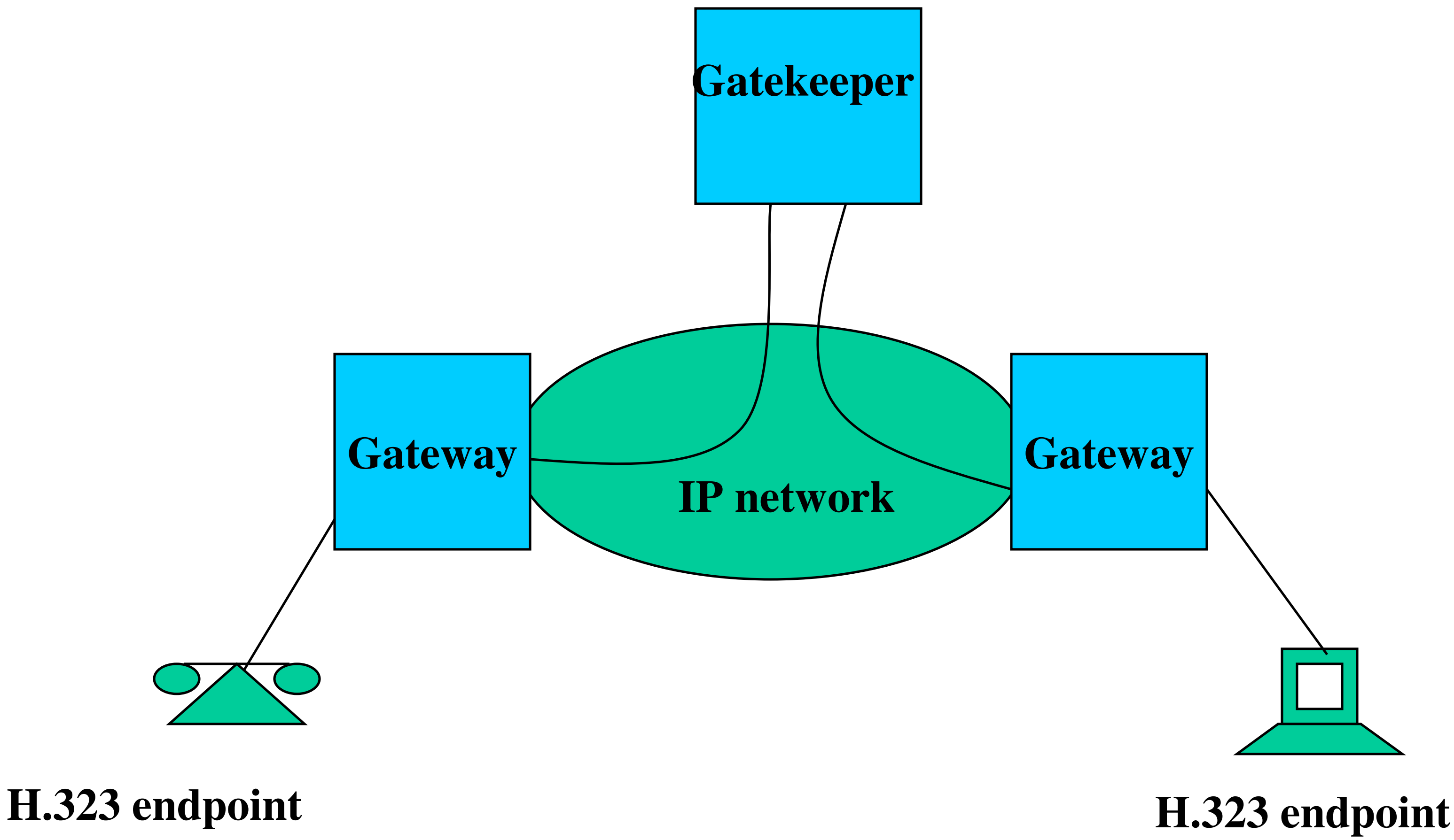
VoIP Protocol Soup



Call Control Signaling Protocols: H.323

- ITU-T defined standard
- Originally developed for ISDN based multimedia services over LAN
- Distributed protocol model
- Consists of
  - Terminals
  - Gatekeepers
  - Gateways
  - Multipoint control units
- Umbrella protocol comprising of several other protocols like H.225, H.245, T.120 etc. defining RAS, capability negotiation etc.
- Binary ASN.1 encoding
- H.323v4 currently implemented everywhere
  - Future H.323v5

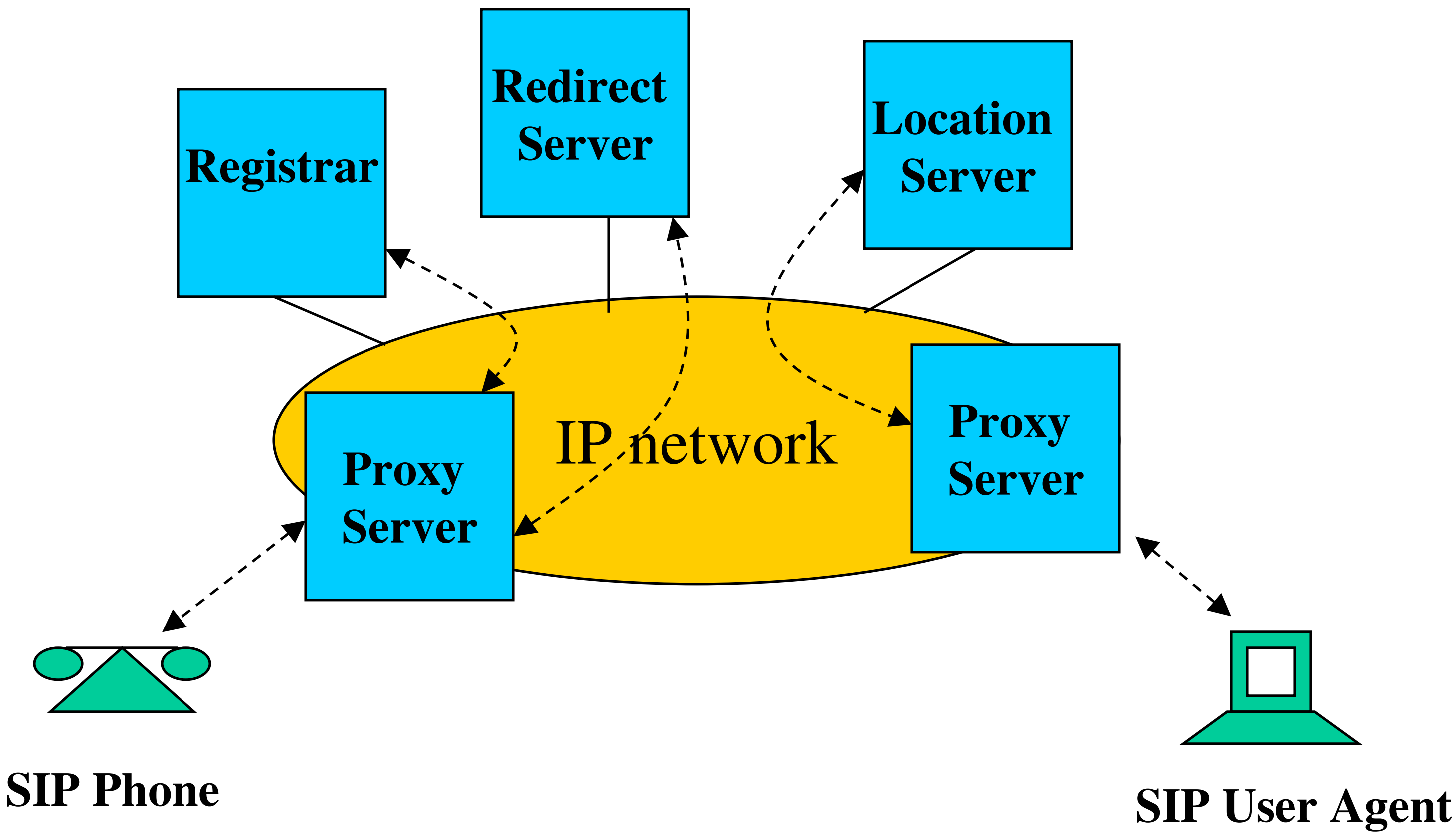
Call Control Signaling Protocols: H.323



Call Control Signaling Protocols: SIP

- IETF RFC 3265 (obsoletes RFC 2543)
- Developed for multimedia services over IP networks based on http model
- Designed to employ existing popular Internet protocols like DNS, SDP etc.
- Distributed model consisting of User agents and Servers
- Text-based implementation is perceived to be simpler, modular, easily adaptable to the www

Call Control Signaling Protocols: SIP



H.323 versus SIP: Reality?

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- ITU protocols more tightly defined; IETF looks for looser working code
- H.323 older and more established; SIP relatively newer but fast catching up
- H.323 widely deployed today; SIP is being widely adopted by large players
- Importance of the Internet and web-based applications increasing
- SIP capable of giving service providers greater control of services, extensibility and interoperability with the www; hence, may eventually win the race
- For a long time however, both these protocols need to co-exist
  - Robust standards must be developed to define interoperability to make things easier

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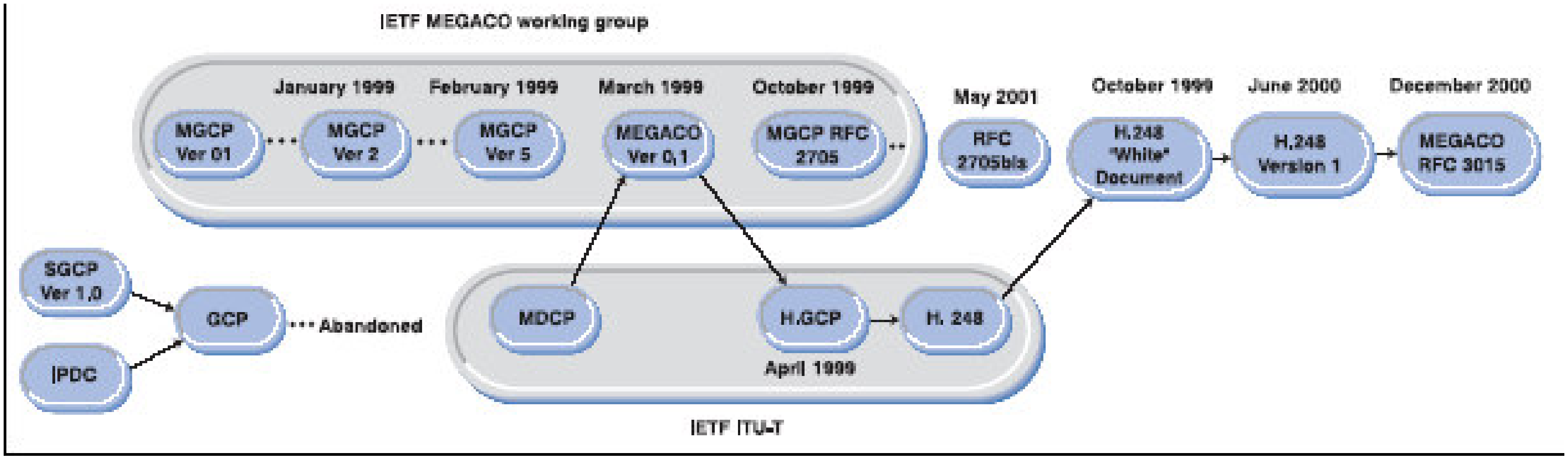
Call Control Signaling Protocols: GCP Evolution



Source: Hughes Software Systems

Call Control Signaling Protocols: MGCP

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- IETF informational RFC 3661
- Provides call control services in a packet network
- Early implementation of Master/Slave protocol
- Consists of media gateways and call agents
  - Call Agents-> centralized intelligent entities handling call control and signaling
  - Media Gateways-> dumb devices handling media
  - Call Agent communicates with Media Gateway via MGCP
- Now a closed effort from standards perspective
  - MGCP implementations do exist today. MGCP variants NCS/TGCP are adopted by Packetcable.

Call Control Signaling Protocols: Megaco/H.248

- Enhances MGCP
- Joint effort by ITU and IETF (IETF nomenclature- Megaco/RFC 3525, ITU nomenclature- H.248)
- Provides call control services in a packet network
- Adopts the Centralized model
- Supports IP/ATM networks
- MGC-MG communication via Megaco/H.248
- Deals with contexts and terminations
  - decouples physical terminations from logical (ephemeral) ones
  - more suited to handling multimedia
- More complete and robust, standard allowing for multi-vendor interoperability

Controller - Controller Protocols: SIP-T

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- IETF RFC 3372
- Defines a framework to interface SIP with ISUP
  - To maintain feature transparency in the SIP network w.r.t PSTN to support IN services not supported in SIP
  - To deliver SS7 information (in its entirety) to some trusted SIP elements
- Integration methods
  - Encapsulation of ISUP within SIP using MIME
  - Translation of ISUP parameters to SIP header
  - Provision to transmit mid-call ISUP signaling messages through INFO method

Controller - Controller Protocols: SIP-T

- Implemented at SIP-PSTN boundary gateways
  - Carried end to end
- SIP-T is relevant in the following scenarios
  - PSTN origination, IP termination
  - IP origination, PSTN termination
  - PSTN origination, PSTN termination with IP transit
  - IP origination, IP termination : SIP-T is not required

Controller - Controller Protocols: BICC

- Development triggered by a need for a packet-based PSTN replacement
- Functional separation of call and bearer signaling protocols in a broadband network
  - IP/ATM bearers in addition to TDM bearer
  - Uses SS7 signaling (with extensions to ISUP)
  - Binding information allows correlation between call control and bearer
- BICC defines three capability sets
  - CS1: supports ATM-based (AAL1/AAL2) bearer
  - CS2: supports IP-based bearer
  - CS3: still in works to support advanced services and interoperability with SIP

SIP-T versus BICC

SIP-T

- IETF defined
- Defined to maintain feature transparency across SIP networks (deliver ISUP to SIP endpoints)
- Packet-based signaling and bearer
  - SIP signaling
  - IP bearer; ATM supported through RFC 3108 (may be some issues, but defined)
- Provides a transition to pure advanced multimedia services based SIP network

BICC

- ITU defined
- Defined to separate call control from bearer (extends ISUP to handle packet bearers)
- SS7 signaling, Packet bearer
  - Network is SS7 (ISUP) signaled
  - TDM/ATM/IP bearers
- Intended for packet-based next-generation network supporting all existing legacy services.

Bearer Protocol: RTP

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- IETF RFC 3550 (obsoletes RFC 1889)
- End-to-end network transport services for multimedia applications
- Services include payload type identification, sequence numbering, time stamping and delivery monitoring
- Control protocol (RTCP) to monitor data delivery
- Can be used with any transport protocol
  - Depends upon underlying transport layer for QoS
  - Applications typically use RTP over UDP

Putting All Ingredients of the Soup together !

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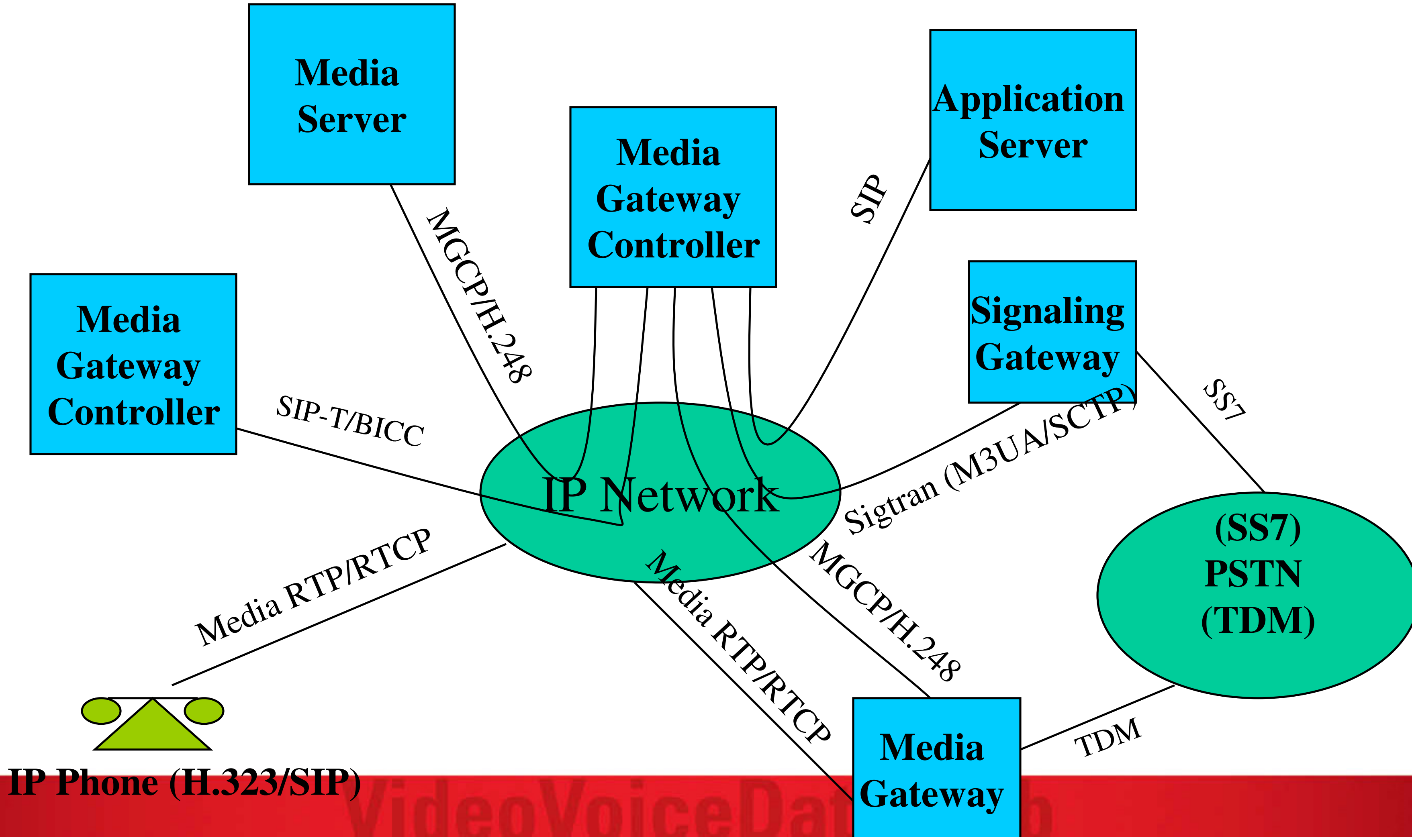
Protocol selection is a strategic decision depending on existing network and future services planned

Ultimately, one winner will make it easy for all !



VoIP Network Performance

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The Key Parameters

- Coding Algorithms
- Echo Cancellation
- Latency
- Jitter/Jitter Buffer
- Packet loss
- Transcoding/Tandeming
- QoS
- Reliability/ Availability
- Quality

Coding Algorithms

- **Compression**
  - **What codec is used and their corresponding bit rates**
    - **Greater the compression, more the encoding delay**
    - **Determining appropriate packetization times and packet length**
  - **MOS score of codec determines perceived quality**
- **VAD and CNG**
  - **At the transmitter**
    - **Detection of voice activity**
    - **Suppression of silence**
  - **At the receiver**
    - **Comfort Noise generation**
    - **Voice playback**

Echo Cancellation

- Echo detection and cancellation
- Availability and echo signal return loss quality
- Adjustments to loudness rating
- Tail length is MG role dependent

Latency/Delay

- Packetization Delay
- Propagation Delay
- Network Processing Delay
- Jitter buffer delay and speech playback
- PLCs add about 5ms delay
- PSTN benchmark for toll quality voice is 150ms RTT (ITU G.114)
  - Delay greater than 300ms is completely unacceptable for toll quality
  - An occasional packet loss is tolerable, but latency beyond 200 ms RT is not.

Jitter/Jitter Buffer

- Jitter = Delay Variation
- Jitter Buffer compensates for jitter on the receiver side
- Jitter buffer size should be optimally chosen
  - Rule of thumb: Jitter Buffer size = atleast 2 x speech frame size
  - Absolute jitter buffer size = end-to-end delay variation + some safety margin
    - Used by Gateways that have more processing power

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Packet Loss

- Packet loss should be below 1% for acceptable quality
- Use Codecs with packet loss concealment algorithms
  - E.g G.729, G.723.1 have built in PLC; add-on PLCs have to be used with G.711 and G.726
- PLC algorithms compensate about 40 ms of missing speech.
  - Delay >40ms & <200 ms, speech is clipped
  - Delay >200ms, speech dropouts
- Packet loss is mostly bursty in nature. Hence, packet loss performance is directly related to packet size, the shorter the better

Transcoding/Tandeming

- Transcoding: Two or more encodings of a signal through different types of non-G.711 codecs separated by G.711 e.g G.726 to G.711 to G.729A
- Tandeming: Two or more encodings of a signal through same types of non-G.711 codecs separated by G.711 e.g G.729A to G.711 to G.729A
- Transcoding increases distortion and delay
- Only one transcode can be tolerated before the network performance drops to unacceptable levels for most combinations of non-G711 codecs

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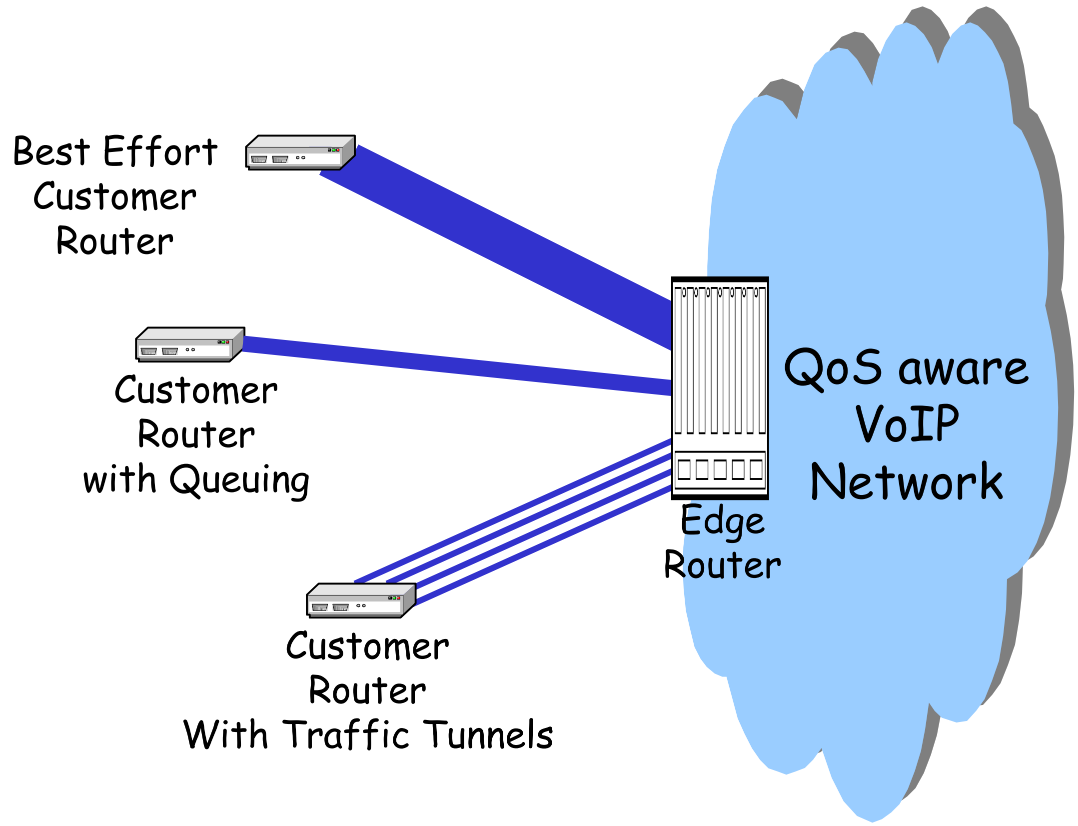
Quality of Service

- Means to prioritize voice packets
- Real time voice packets receive higher priority than non-real time data packets
- Helps improve performance by decreasing delay/jitter for voice packets
- Significant delay/jitter events can be avoided only by implementing a proper QoS Strategy

QoS Strategy

- **Best Effort**
  - **A class of service in which the network provides no guarantees to the edge equipment**
- **Prioritized Queuing**
  - **Differentiation in the queuing of traffic for various classes of traffic**
  - **Assigns a priority or classification to every IP packet**
  - **Packets are sent in order of priority**
- **Traffic Engineered Tunnels**
  - **Constraint-based (traffic sensitive) connection-oriented paths through a routed network**
  - **MPLS Label Path**
  - **ATM VC**



Speech Quality

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Speech quality important for

- Monitoring/fault-finding
- Service level agreements
- Optimisation of network

Quality will remain an issue so long as bandwidth or processing power are limited

- e.g. mobile, leased capacity

Factors that affect quality

» Background noise
» Silence suppression
» Low bit-rate coding
» Errors (mobile, packet)
» Delay
» Echo
» Handsets/access network

- Quality measures: MOS; PSQM (Perceptual Speech Quality Measure); PAMS (perceptual analysis measurement system); PESQ (Perceptual Evaluation of Speech quality)
- End-to-end speech quality is the key measure of voice QoS

The Biggest Challenge

The VoIP Network Cloud

The VoIP network must:

-**provide a customer's service preferences anywhere**

- **securely**

-**adjusting to the constraints of the networks and access methods used: Wireline, Wireless, 3rd Party, Customer Owned, or Public Internet**

Clearly a list of codecs, packet sizes, loss rate and jitter targets are needed to ensure voice quality to a defined level of acceptability.

VoIP Network Security

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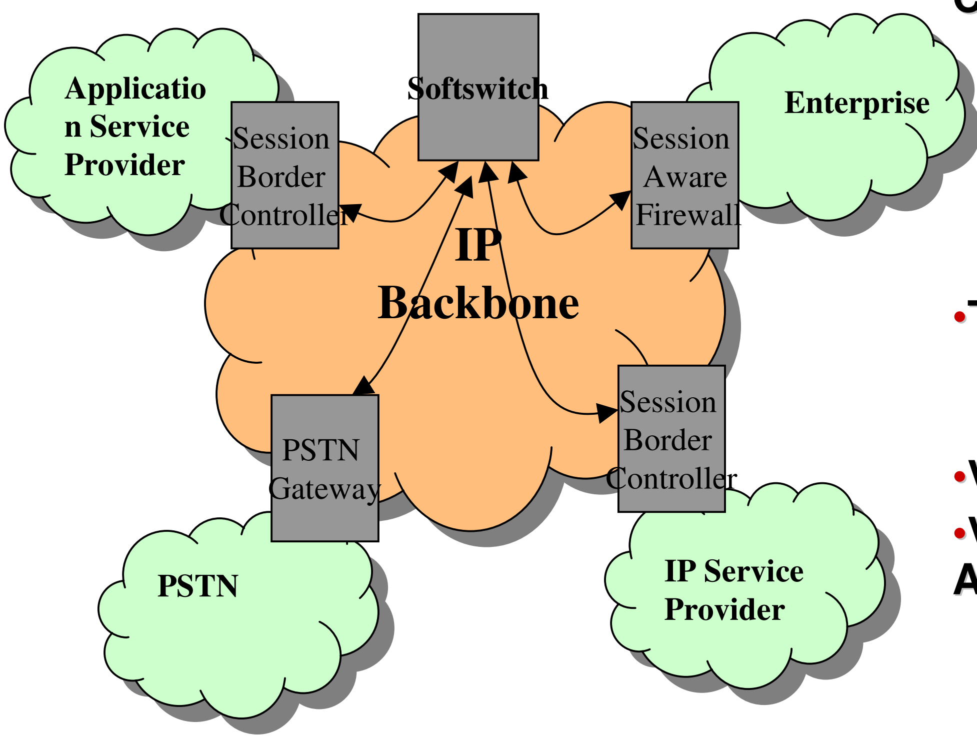
New IP Requirements

- Viewing packets as part of sessions
- Policies are required for Sessions
- New IP Services are enabled to handle this
  - Routing sessions between different networks, carriers and domains
  - Session packet flow anchoring
  - Detect failures and reroute
  - Usage based Billing/reporting at session flow level
  - Session aware borders for security

Session aware firewalls

- Two way VoIP communication impeded by NAT/NAPT
  - Signaling messages can be exchanged on defined port
  - Bearer messages are a problem
- Special tags in SIP message to permit two way communication
- Simple Traversal of UDP messages over NAT (STUN)
  - Creates NAT awareness in Clients
  - To modify SDP messages

Session Border Controller



- VoIP Firewall Traversal Solution for Carrier to Carrier Peering
  - Integrated SIP Application Layer Gateway (ALG)
  - Modify signaling traffic to accommodate NAT/NAPT
  - Dynamic 'pinhole' opening/closing
- Topology Hiding
  - Provide an address normalization boundary
- VoIP Media Anchoring Solution
- VoIP Session QoS / Service Level Agreement Solution
  - Per session based policing
  - Guaranteed service in congested environments
- VoIP Session Admission Control Solution

VoIP Regulations

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Regulations

- Numbering Services
  - Rate Centre Association of Numbers
  - Impact on Number Conservation
  - Number Portability Compliance for VoIP providers?
- Information service versus Telecommunications service
  - Access charges at Origination and Termination points?
- CALEA
  - Requires North American telecommunications carriers to modify their equipment, facilities, and services to ensure that they are able to comply with authorized electronic surveillance.
  - Similar requirement for the VoIP?

VoIP Network Economics

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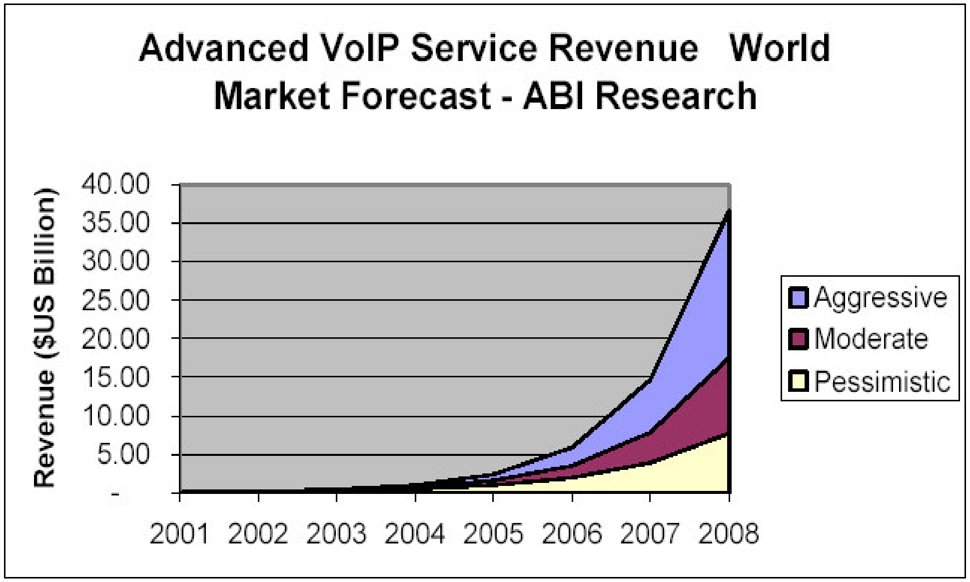
Is VoIP economical?

- What Service Providers want
  - Decrease expense
    - Lower capex on new infrastructure elements introduced
    - Lower opex on maintenance of existing infrastructure
  - Increase ARPU
    - Increase talk-time and use of Value added services (Centrex)
  - Ease of feature incorporation
  - Ease of Application development and innovation
  - Distributed architecture, fewer elements

Deployment Numbers suggest the same

- **In 2003, IP accounted for the majority - 58% of network traffic - TIA**
- **International VOIP calls grew from 0.2% of telephone traffic in 1998 to 10.4% in 2002 - Broadwatch News Analysis**
- **Total Internet protocol (IP) revenues expected to grow in 2004 by 7.8 percent, achieving a total of $13.9 billion – TIA**



Source: ABI Research

Major VoIP Providers

Vonage - 70,000 subscribers

Free World Dial-up - 80,000 subscribers

Yahoo BB - Over 3 Million subscribers

FastWeb - 400K subscribers

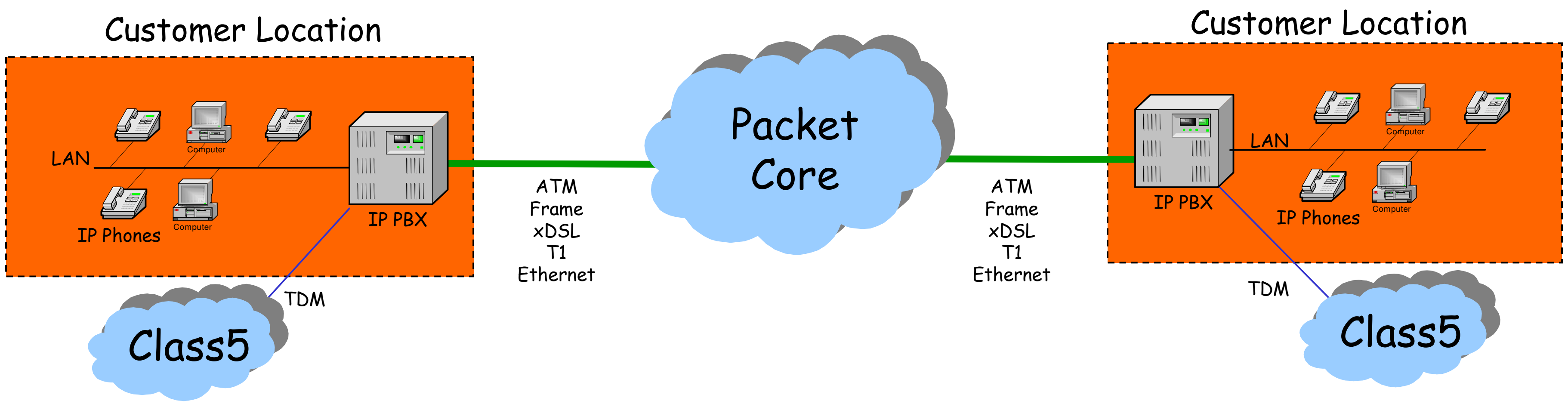
VoIP Network Applications

Application Scenarios

- IP PBX
- IP Centrex
- Enhanced IP Telephony
- Class 4 Replacement
- Class 5 Replacement
- *And more…..*

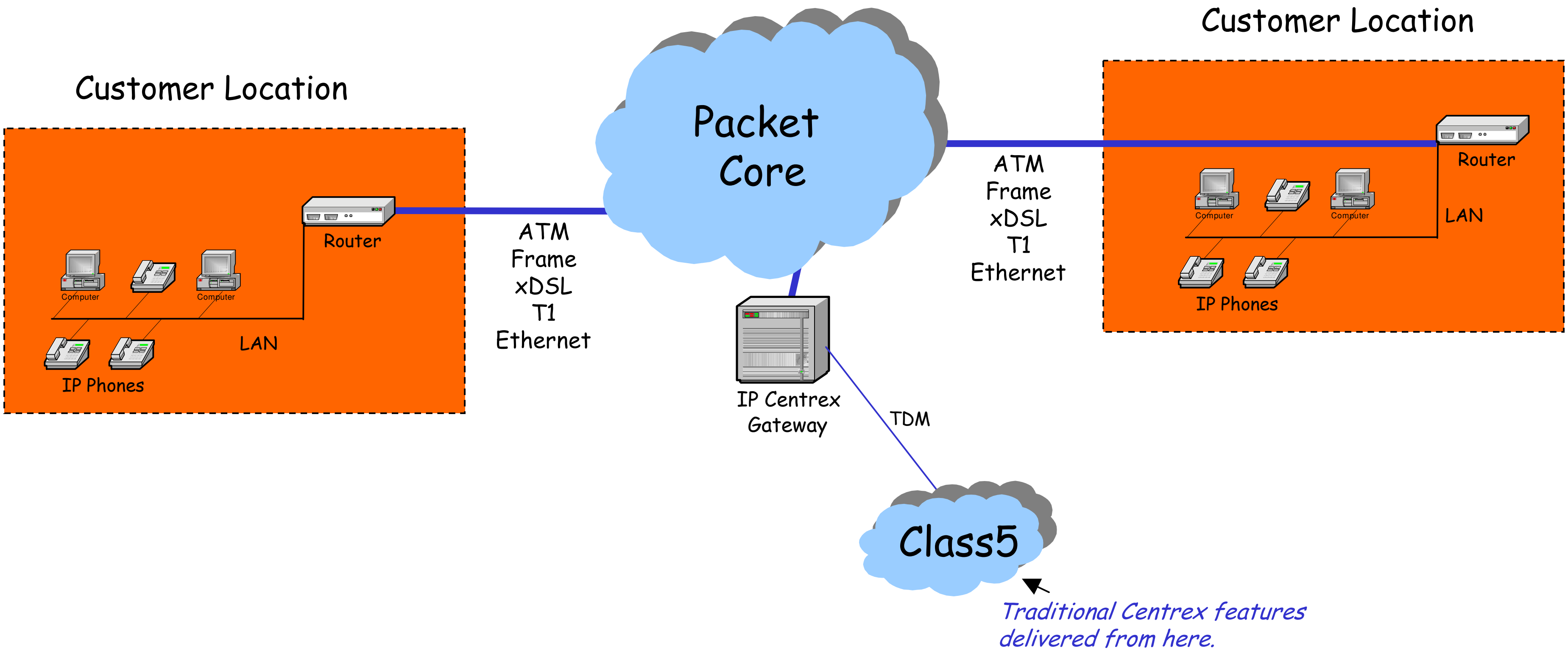
IP PBX

- **A CPE based IP telephony service that replaces a traditional TDM PBX**



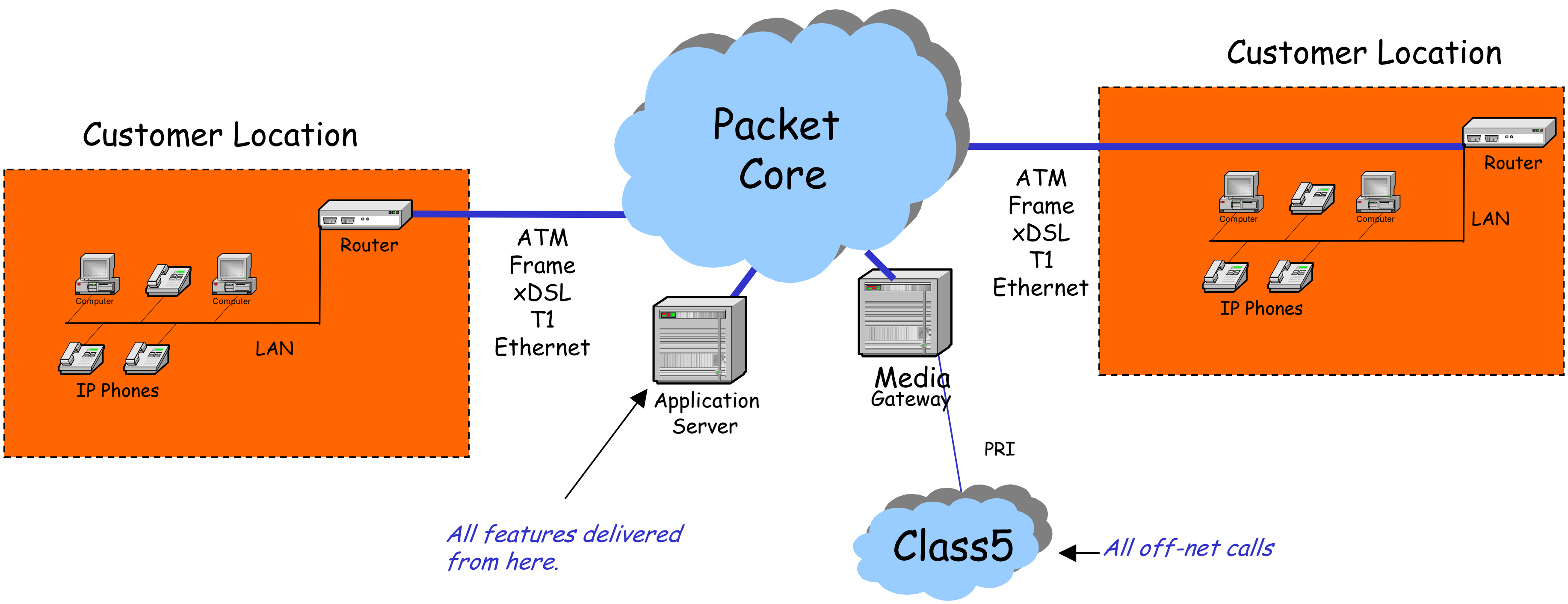
Basic IP Centrex

- **A network based IP telephony service that leverages a traditional Class 5 based Centrex service.**



Enhanced IP Telephony

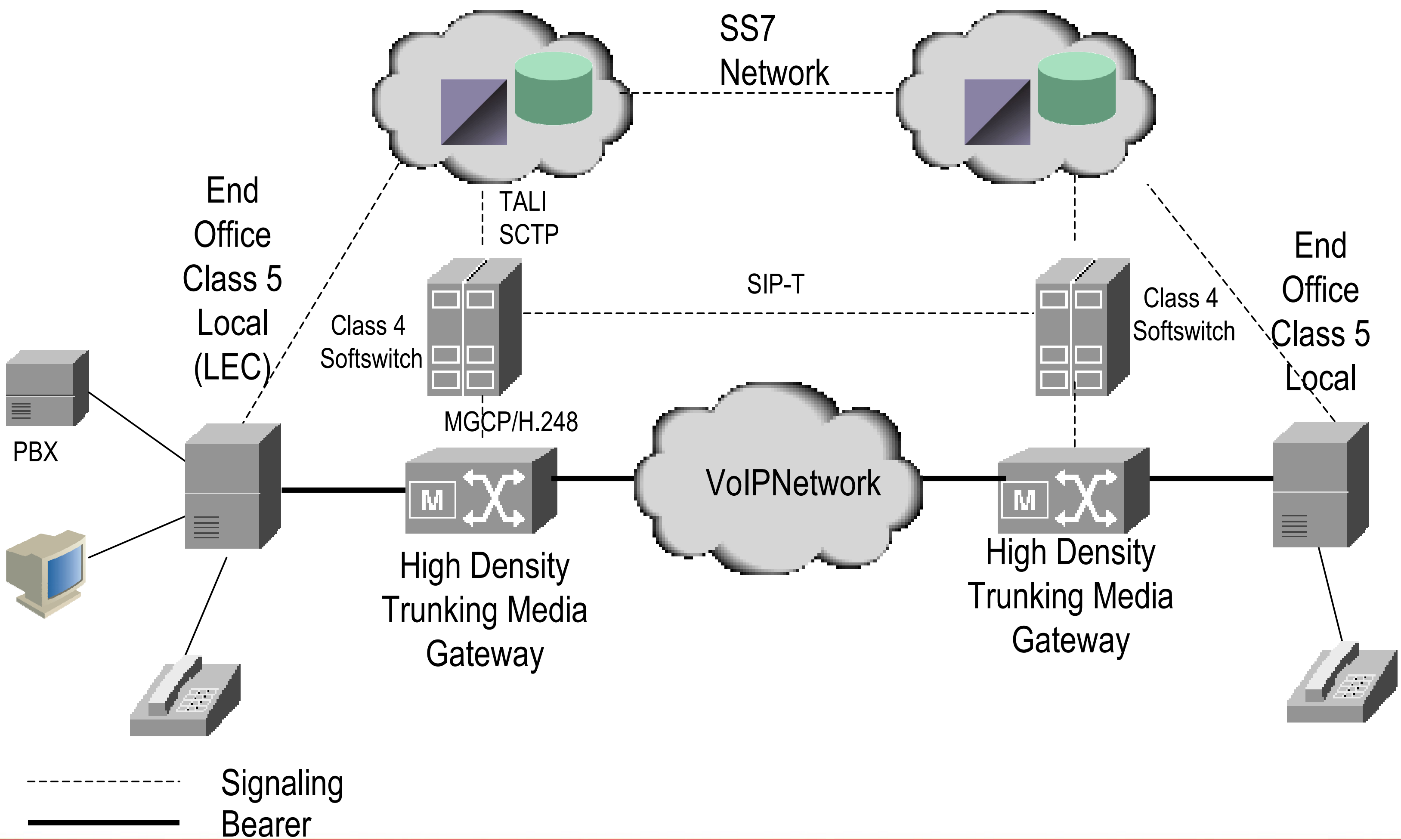
- **A network based IP telephony service that provides multi-media voice over an IP network, in addition to basic Centrex features.**



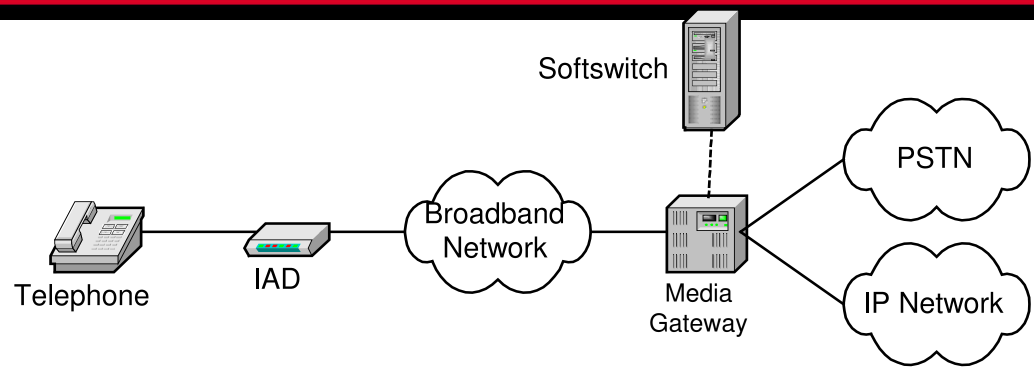
Class 4 Replacement

- Scenario
  - ILECs, CLECs, IXCs, Large Corporations
- Benefits
  - By-pass traditional long distance toll network (Class 4) carriers and their per-minute usage rates and run their voice traffic over IP networks for a reduced cost.
  - Lower costs with higher bandwidth efficiency
- Issues
  - Traffic engineering of IP network for PSTN QoS
  - Migration from Circuit to Packet-based Network

Class 4 Replacement



Class 5 Replacement



Scenario

- Out of Region and "Greenfield" deployments

Benefits

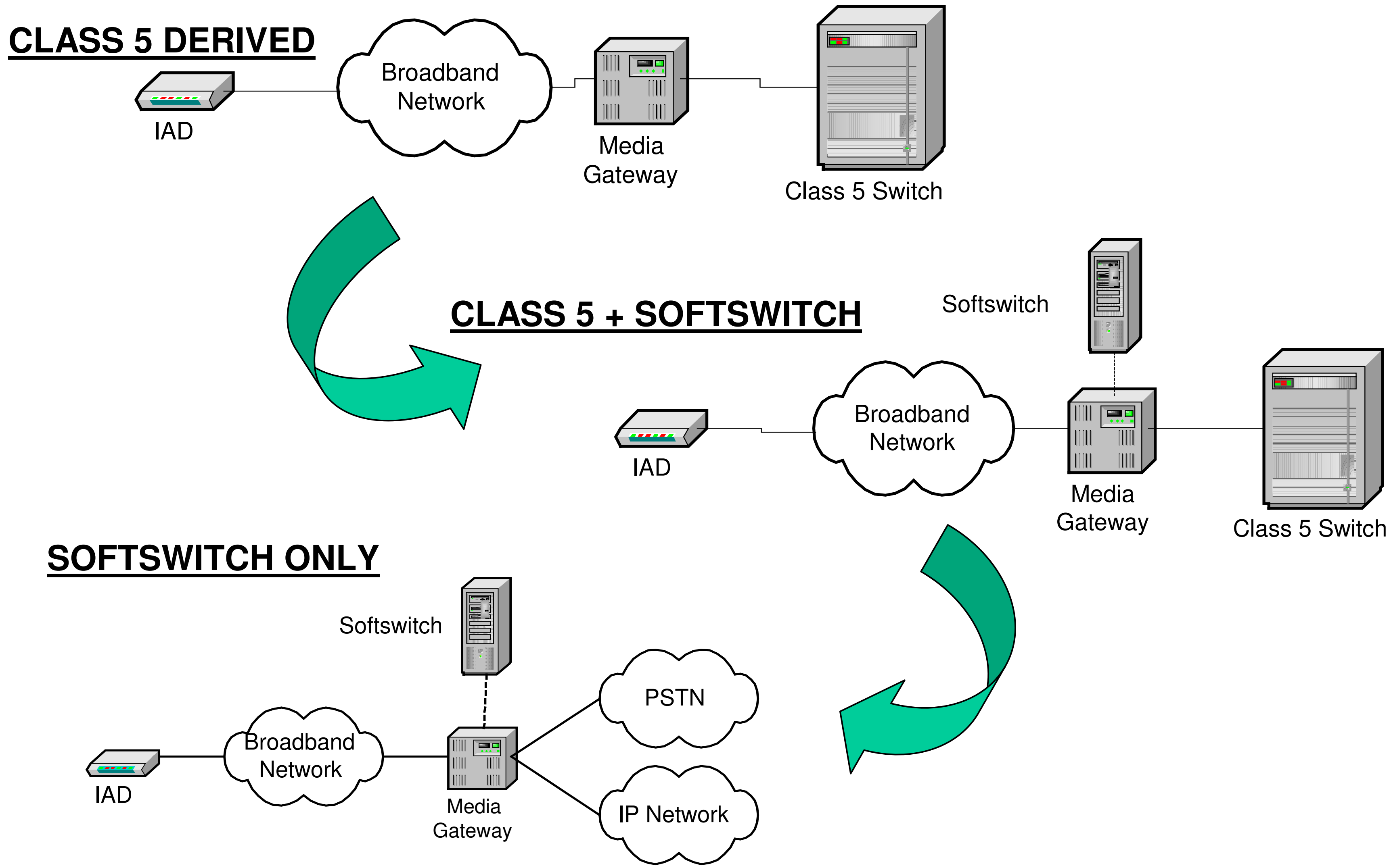
- Flexibility - Enable Rapid Deployment of New Services
- Distributed Architecture rather than Hierarchical Class Model

Issues

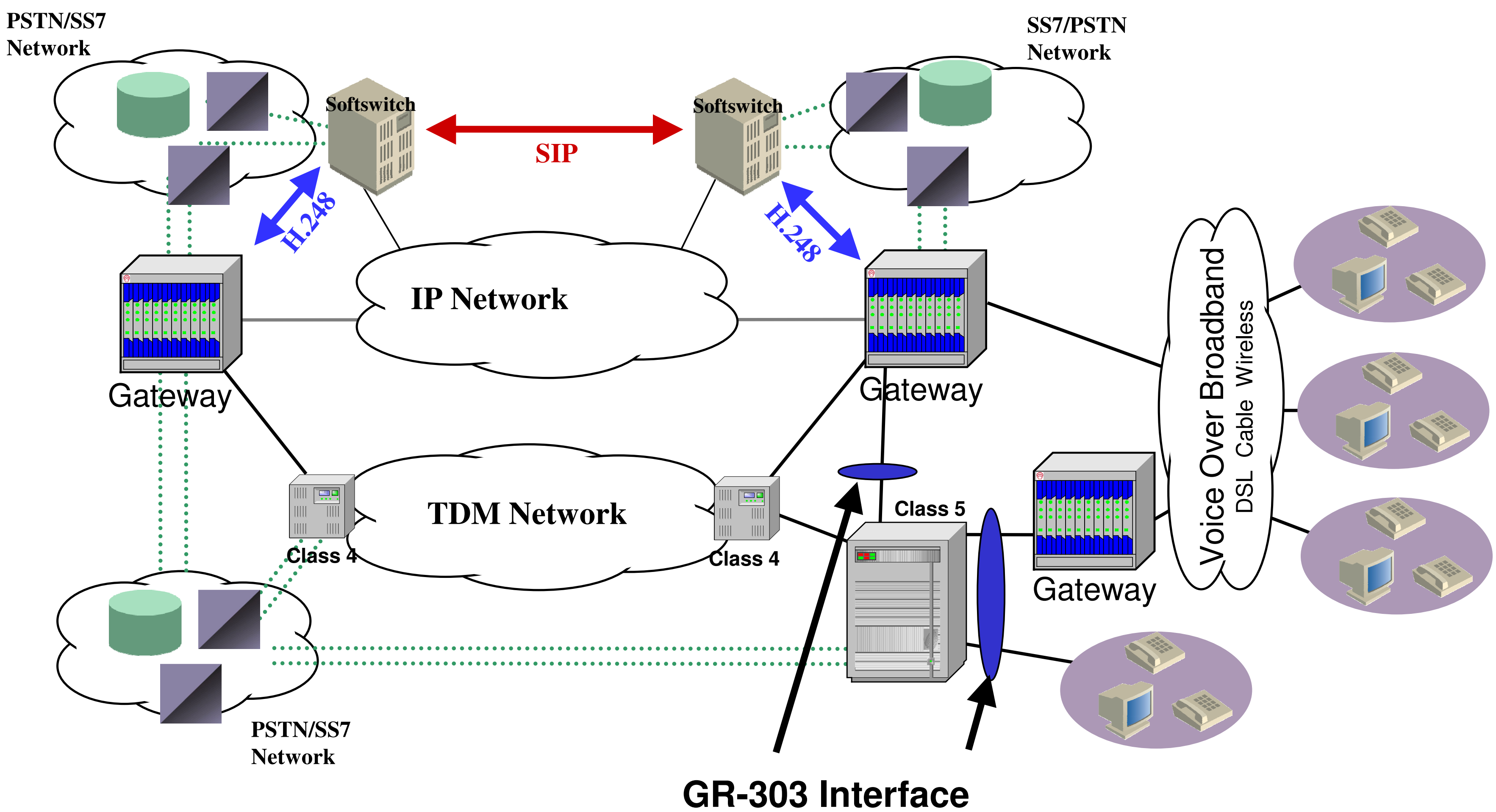
- Maturity of softswitch technology
- Ability to support all legacy systems supported by a Class 5 switch

Evolution to VoIP network

Conservative Migration and Evolution



Voice over Broadband Network Architecture



Evolution of VoIP - Generic View