Converged Services and a New Generation of Networking

... for discussions only ...

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Overall Presentation Outline

- Convergence of Communications
  - VoIP, IPTV, Streaming media, etc.
- Architecture for New Generation of Networking
- Wireline and Wireless Broadband Access
- Multimedia Traffic Transmission Techniques
- Revenue Model and Research Topics
- Q&A and Open Discussions

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Outline of this Section

- Convergence of Communications
  - Legacy Communications Services
  - Today’s Communications Services
  - Emerging Communications Services
- Legacy Voice ➔ Voice over IP (VoIP)

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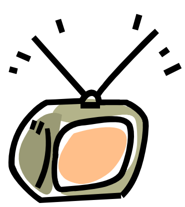
Legacy Communications Services

- Voice (DS0/64Kbps)
- Video (Analog)
- Data (19.2 Kbps)
- Narrowband Pipes
- Analog Pipes

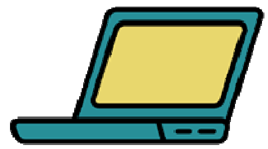


4 MHz Processor
64 KB RAM
264 KB Disk
Expensive/Shared PC
Asymmetric Bandwidth

Today's Communications Services

- Voice over the Internet Protocol (VoIP; HD)
- IP based TV (IPTV; SD/HD) Services
- Electronic Mails (Emails)
- Messaging: IM/SMS
- 100 Mbps to Home (FTTH)
- One to 5 Mbps Wireless
  - LTE and M-WiMax
- Clouds

3 GHz Processor
4 GB RAM
500 GB Disk
Cheap/Portable PC
Symmetric Bandwidth



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Emerging Communications Services

- Streaming NG/3D Media Service
- Blended/Converged Services
- Multi-Screen Mobile Culture
- Evolved Social Networking Services
- Open Sourcing & Global Development
- Consumers are the KINGs / QUEENs
- Resiliency through Distribution
- COTS & Virtualization
- Broadband Digital Pipes

Multi-Core Multi-GHz Processor
16 GB or more RAM
Multi-TB Disk
Wearable/Embedded PC
Asymmetric Bandwidth (CGC)

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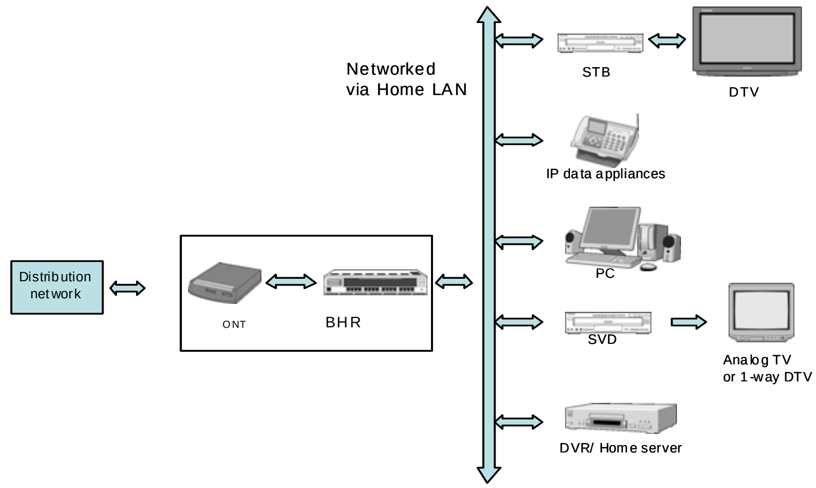
Complexity of Home Networks



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… Voice Service Evolution …

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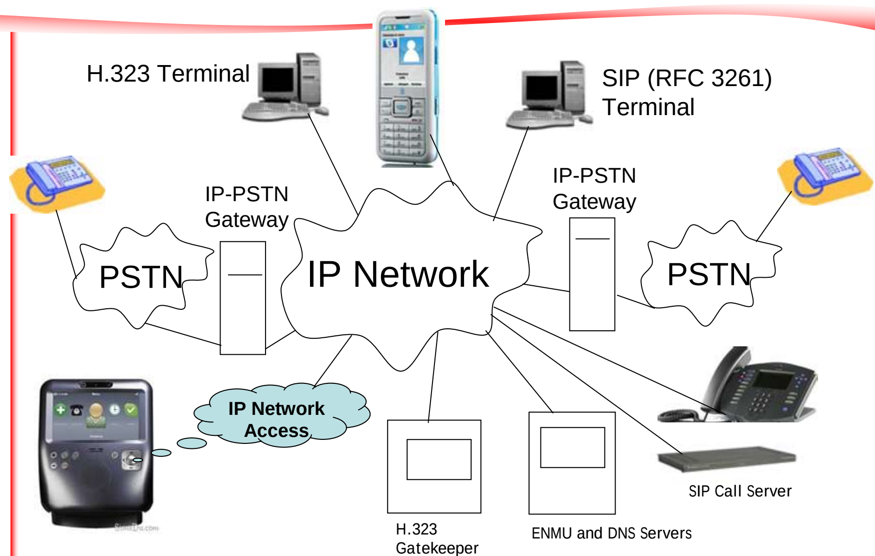
Voice over IP Network Elements



VoIP Protocol Stack

Call Establishment and Control					
	Presentation				
Addressing	Audio Codec: G.723.1, G.729, …	DTMF			Addressing
RAS	SIP	RTP/RTCP	H.245	Q.931 H.225	DNS
UDP (RFC 768)			TCP (RFC 793)		
Network (IPv4-RFC 791 or IPv6-RFC 2460/4294)					
Link					
Physical					

Bandwidth Estimate for Voice over IP/Ethernet

- Voice payload length = codec bit rate * packetization delay:
- Overhead for VoIP/Ethernet:
  - RTP (12bytes), UDP (8bytes), IP (20bytes) = 40 bytes
  - LLC/SNAP (8 bytes), MAC layer header (14 bytes), preamble (8 bytes), MAC FCS (4 bytes)

4		12	8	20	8	14	8	12	Bytes
FCS	Voice Payload	RTP	UDP	IP	LLC/ SNAP	MAC Header	Preamble	InterFrame Gap	

$$\text{Bandwidth} = \frac{(4+\text{ Voice Payload } + 40 + 8 + 14 + 8 + 12) \times 8}{\text{Packetization Delay}}$$

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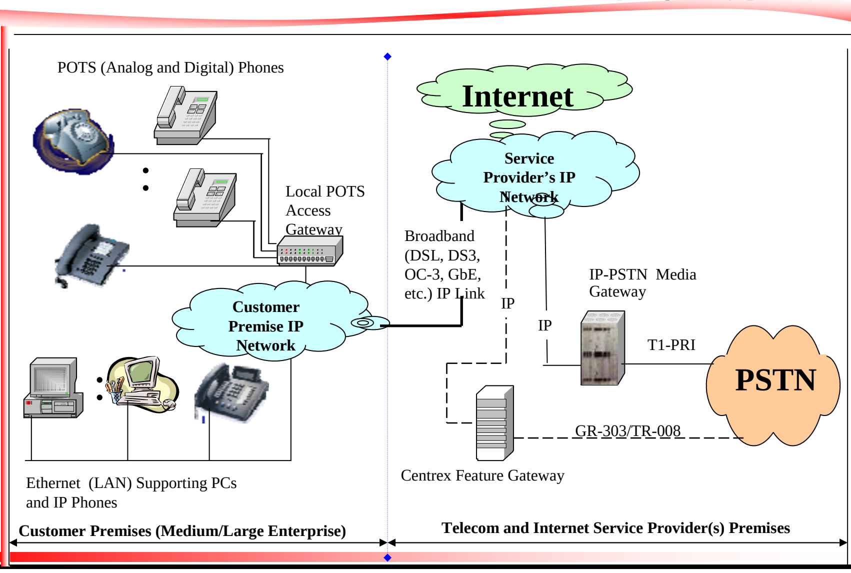
VoIP Based Centrex Service (legacy)



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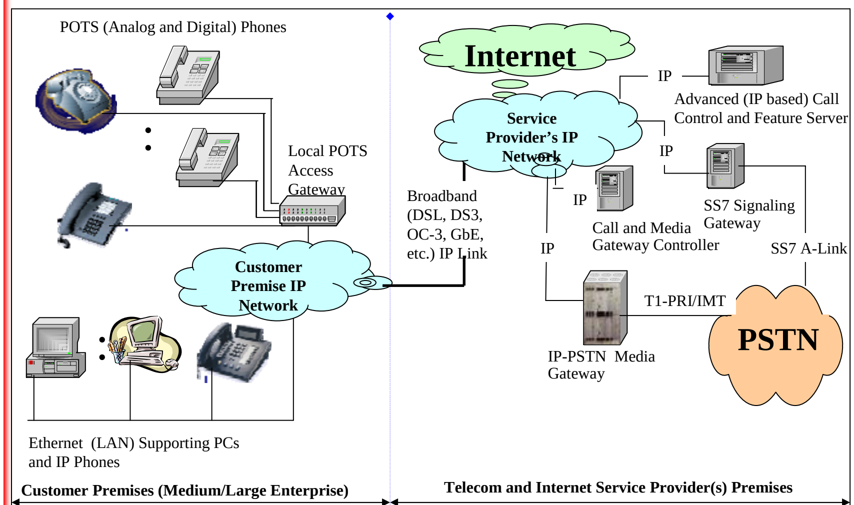
VoIP Based Centrex Service (Emerging)



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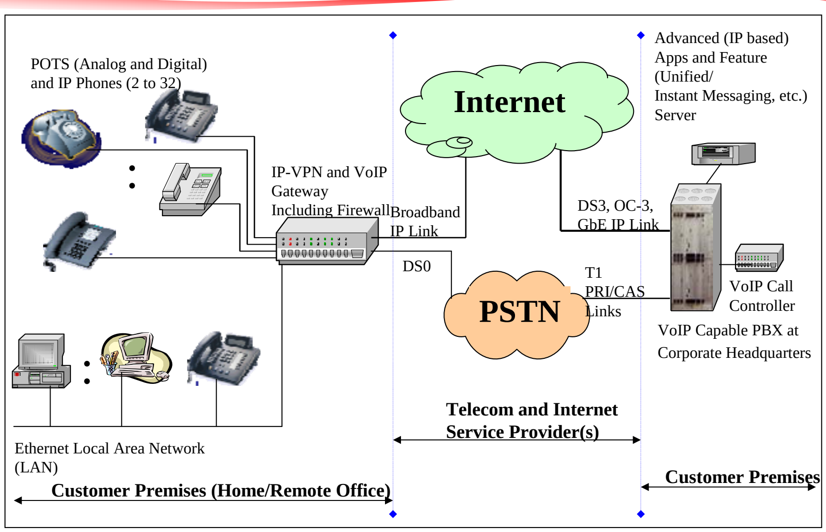
VoIP for Remote/Traveling Tele-workers



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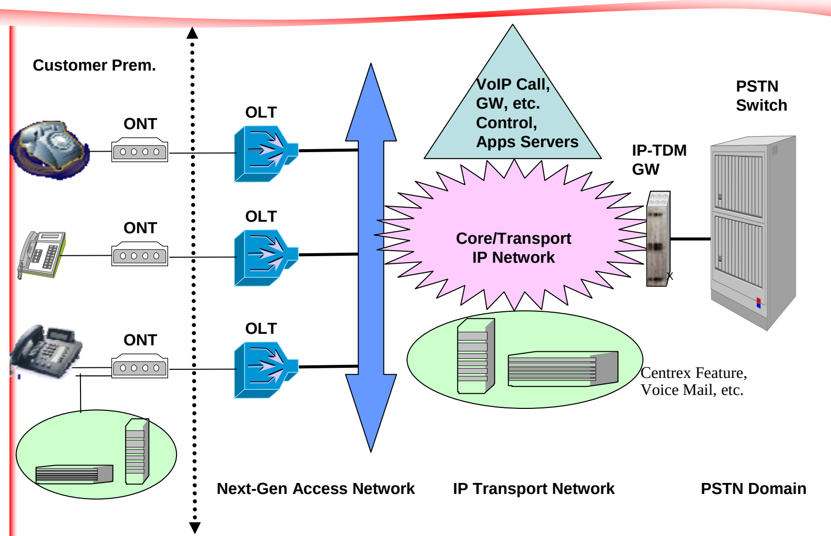
VoIP for Fiber-based Access Lines



VoIP Access: Issues and Solutions

- Service during power outage, and POE support
  - Dual power supply, IEEE 802.3af (www.ieee802.org/3/af/ , www.poweroverethernet.com/) implementation
- Regulatory and safety concerns for e.g., E911 call routing with location Info to PSAPs
  - User Profile and Network Server based Management of Location identification is being explored
- End-to-end traffic and security management
  - Both layer-2 and -3 issues need to be addressed
- Modular/Structured Wiring, and Segmentation to support VLANS, QoS, etc.
  - Wiring and LAN switches may need to be upgraded
- Seamless delivery of high-quality service

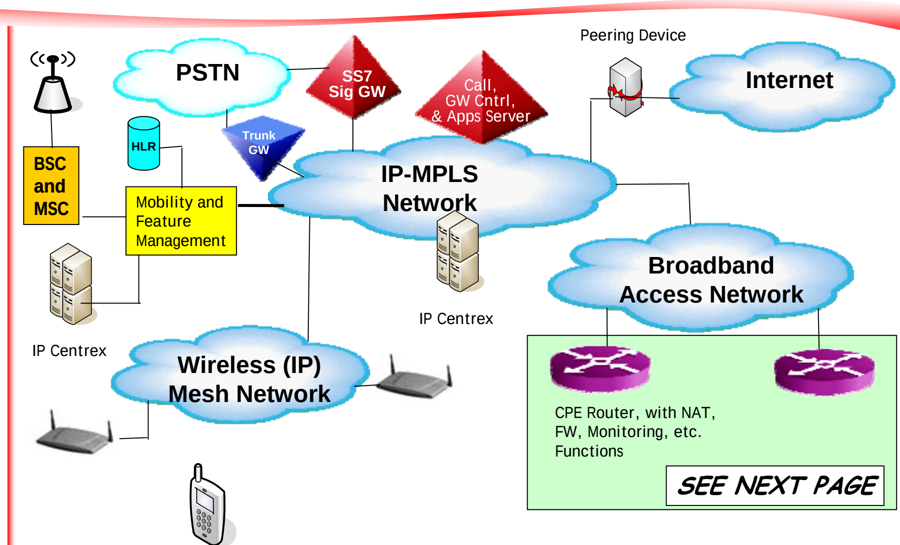
VoIP over Wireless Access (../1)

- VoIP over WiFi
  - SIP clients can access the service once communication to the WAP (wireless access point) is established
  - Additional security & signal boosting may be required
- VoIP over Broadband Wireless Access
  - This is same as the VoIP support over broadband wire-line access, except that the CPE or IAD now has broadband wireless access (e.g., IEEE 802.16) to the networks
  - These challenges consist of maintaining proper strength of the signal in presence of interference, fading, failure of electric power supply, adverse atmospheric conditions, etc.

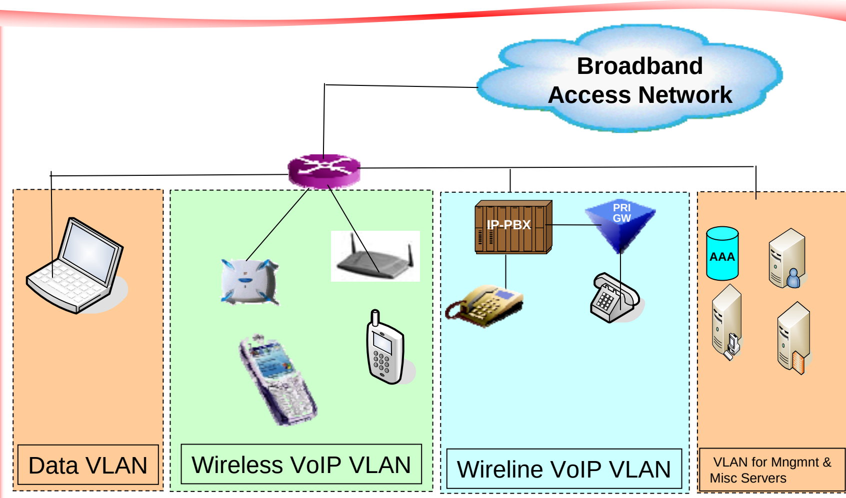
VoIP over Wireless Access (../2)

- VoIP over Wireless phones
  - Dual (VoIP over WiFi and cellular; SIP & CDMA clients) mode phone
    - Signaling and media gateways to both (circuit switch based) Wire-line and wire-less networks are required
      - Signaling (SIP)/{TCP, UDP}/IP/802.11x & Media(G.729b)/RTP/UDP/IP/802.11x
      - Voice & signaling of CDMA/RF
    - Suitable for service providers who have (or can support) both wireline and wireless VoIP infrastructure
  - Data Connection (**EV**olution **D**ata **O**ptimized)
    - Average access bandwidth of 500 Kbps is sufficient to support multiple simultaneous VoIP sessions
    - Seamless handover may become an issue unless it is addressed carefully
    - Service theft may become issue unless proper billing or blocking mechanism is activated

VoIP over Wireless Access (../3)



VoIP over Wireless Access (../4)



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Wireshark Capture of SIP Messages (partial)

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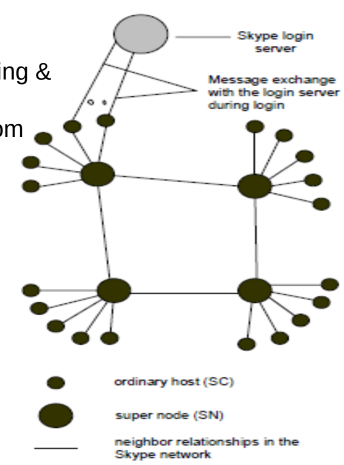
Skype Voice Service

- Skype is a peer-to-peer (P2P) Application
- Uses Proprietary Protocols for both Signaling & Media Traffic Exchange
- Attempts to use TCP or UDP with a Random port first
- If that Fails, it tries HTTP and HTTPS ports (TCP port 80 and 443)
- Almost all Packets are Encrypted using 256-bit Advanced Encryption Standard (AES) Technique
- Maintains Flow of Symmetric Traffic (does NOT use Silence Suppression)



JNSM, Vol.17, No.1-2, Mar.-June 2009, P.53 and
http://www1.cs.columbia.edu/~salman/publications/skype1_4.pdf

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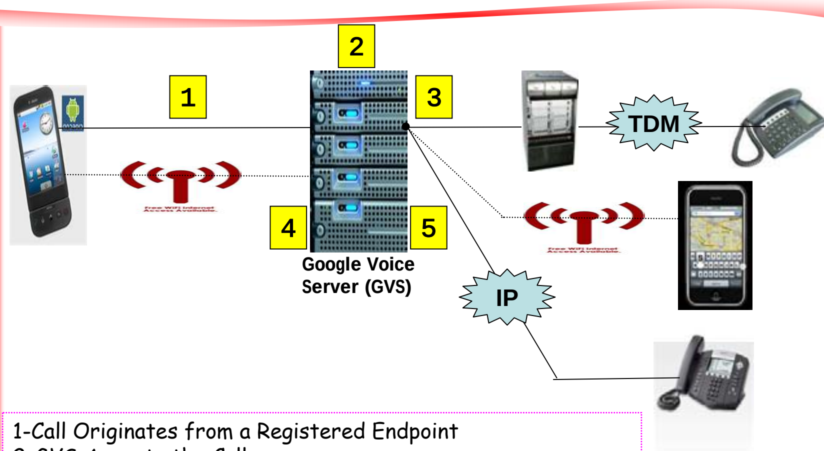
Google Voice Service



1-Call Originates from a Registered Endpoint
2-GVS Accepts the Call
3-GVS Forks the Call
4-Subscriber Accepts the Call via One Endpoint/Device
5-GVS Bridges the Call Legs & Activates Subscribed Features

Google Voice/Talk Service

- As of June 2009 Google registered a million or so phone numbers in preparations to launch Google Voice beyond a private beta. Google plan to support the the following features:
  - ***Call Routing,*** *with* Google Voice number as a primary number, and calls (from individuals or groups) to that number can be routed to cell phones, landlines and voice mailboxes
  - ***Call Screening,*** a user **has Four Options** on what to do with an incoming call (Caller's name is spoken during Ringing))
    - answer, send to voicemail, send to voicemail while listening to the message being left, or answer and record the conversation about to happen
  - **Voice-Mail → Email,** Google Voice can transcribe voicemails and send them to the user via ***email*** or text messaging (audio files of voicemail are saved for online access)
  - **Switching** (using the Star Key on the Phones' dialpad) between calls without interrupting the current call; user can decide what to do with the current and incoming calls during the conversation

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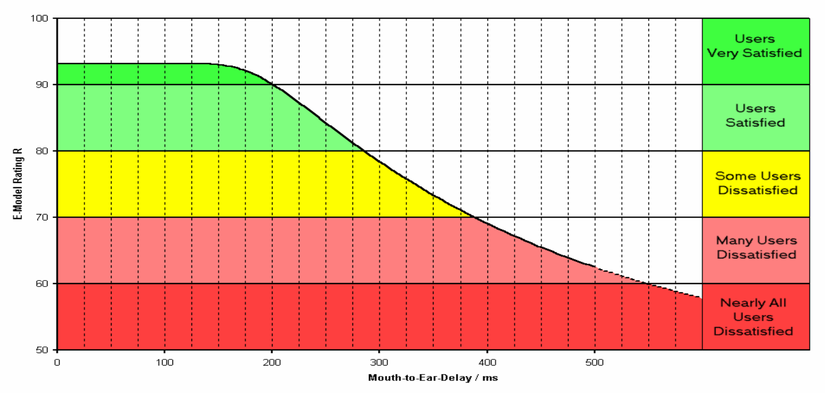
Voice Quality Degrades with End-to-End Delay (E-Model)



ITU-T G.114 recommends that for VoIP the end-to-end delay of less than 150 milliseconds is mostly acceptable, 150-400 milliseconds maybe acceptable and more 400 milliseconds is not acceptable.

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M2ED for Free VoIP Services

- **Skype** had the **best** result followed by MSN and **Yahoo was a distant third**
- Mouth-to-Ear Delay (**M2ED**) for **Skype** service is close to **90 ms**
  - For **Google-Talk/-Voice** the M2ED is **109 ms**.
  - For **Yahoo** the M2ED is **150 ms**
  - For **MSN** the M2ED is **180 ms**

http://forum.skype.com/ http://www.skypestats.com

Source: http://www1.cs.columbia.edu/~salman/skype/index.html

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Factors Affecting the Quality of Experience of a Service

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- Nodal Quality of Service (QoS)
- Link-Level QoS
- End-to-End QoS and Service Level Agreements (SLA)

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Nodal QoS via Packet Prioritization

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Type of Information	**Emission Priority**	**Discard Priority**	**Comments**
~ Urgent and Important	Low	*Mostly* Non- Discardable; (occasionally set Loss Priority, LP=0)	Session Level ***Control*** and ***Signaling*** Traffic
Urgent and Important	Medium	Non-Discardable; (Loss Priority, LP=0)	Network Management and Control Traffic
Urgent and ~ Important	High	Discardable (Loss Priority, LP=1)	***Bearer*** or Media Traffic, e.g., Voice or Speech Signal

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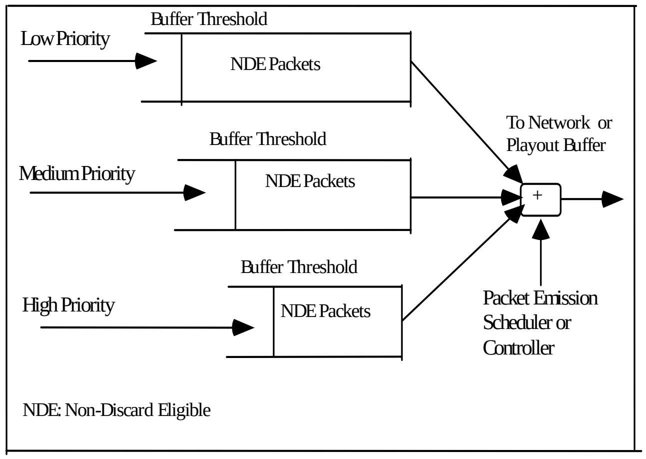
Nodal QoS via Packet Prioritization (…/2)



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Nodal QoS via Packet Prioritization (…/3)

•Let {P_{loss}, ρ, *MTU*, Ca^2, Cs^2}={10^{-6}, 0.95, 128, 3.24, 0.60}, and using the previously cited formulations, the buffer size become approximately 50 Kbytes, as shown below.

$$Q_{size}(Bytes) = \frac{[\ln_e(10^{-6}) - \ln_e(0.95(1-0.95))]\,x128}{\gamma}$$

$$\gamma = \frac{2\,x(0.95-1.0)}{(0.95\,x3.24)+0.6} = \frac{-0.10}{3.678} = -0.02719$$

$$Q_{size}(Bytes) = \frac{[\ln_e(10^{-6}) - \ln_e(0.0475)]\,x128}{-0.02719}$$

$$Q_{size}(Bytes) = \frac{[-13.81551 + 3.047]\,x128}{-0.02719}$$

$$Q_{size} = \frac{-10.76851x128}{-0.02719} = 50KB$$

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Nodal QoS via Packet Prioritization (…/4)

- This 50 KB (Kilo-Bytes) of buffer space is equivalent to 270 msec of *maximum* delay on a T1 (1.544 Mbps) link
- To minimize the maximum queueing delay, the *network design* should consider *minimizing* the "number of *active* nodes crossed" from source to destination
- Consequently, the concept of ***virtual (private) networking*** or VPN comes into picture

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VoIP Traffic Engineering

http://www.erlang.com/calculator/

http://www.erlang.com/voipselectmanual/

- **B**usy **H**our is any "3600-second" or 1- Hour time duration when traffic volume is the largest
- **C**all **A**ttempt is any attempt to achieve a connection
- Busy Hour Traffic or BHT = [(BHCA x AHT) / 3600]
- Busy Hour Call Completion or BHCC = [BHCA x ASR]
- AHT is the **A**verage **H**olding **T**ime
- ASR is the **A**nswer **S**eizure **R**atio (varies from 55% to 75%)

$$BHCA = \frac{[Network.Calls.per.Secx3600]}{Total.NO.of.Subscribers}$$

$$Offered.Load(Erlang) = \frac{Total.No.of.Subscribersx(BHCAxAHT)}{3600}$$

$$NO.of.Ports.\mathrm{Re}qd. = [Offered.Load(Erlang)]xBlocking.Factor$$

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ETE QoS and SLA

- Assuming that the availability of all Nodal and Transmission Elements is independent, we can determine the *Service Availability* as follows:

$$[ServiceAvailability]_{Tier-1} = \prod_{i=1}^{N} \frac{MTTF.E_i}{[MTTF.E_i + MTTR.E_i]}$$

- If the network consists of M-level of tiers (hierarchy), and the availability of each of these tiers is independent, the overall end-to-end service availability (ETE-SA) is:

$$[ETE.SA] = \prod_{i=1}^{i=M} (SA)_{Tier-i}$$

Source: A. Conway & B. Khasnabish, "End-to-End Network Reliability Modeling of Enterprise VoIP Services," NOMS-06, Vancouver, BC, Canada, April, 2006.

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Costs for QoS and SLA

$$Profit = [Revenue - AllCosts]$$

• Costs include Fixed Costs, Operations & Engineering Costs, Regulatory, and Technology Introduction (including Training) costs

$$ProfitMargin(percentage) = \frac{Profit}{Total Revenue}$$

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… Apps that Generate MM Data …

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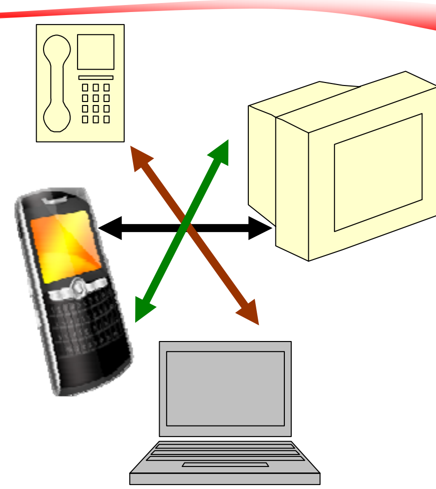
Convergence of Devices

- Phone Call and Emails over TV
- TV/VOD and Emails Services over Cell-Phone/PDA
- Any-media services in a Laptop/PC



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What's Happening Today ?

- Digitize the Contents & Communications
- Personalize and Customize the Services
- Multimedia Focused Interactions
- Interworking of Different NGN Interconnection Technologies
- Budgeting Impairments for Different Segments of the Interconnect
- Seamless Support of Domestic, Regional and Global Variants of Services
- Service-Specific Policy Design & Enforcement
- Service-Specific Security, Tests & Certification

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What Will be Happening in Future ?!

- Apps for Any Services in Any Device from Any Provider (Globally)
- More Machine-to-Machine and Mobile-to-Mobile Communications (in *embedded* fashion)
- Remote and Automated Health Care Initiatives
- Green Initiatives (Environmental Awareness)

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Users' View (as perceived by BT)

- More than 1.5 bn internet users
- 20+ bn web pages on 150+ m. domains
- $3.5+ bn $ online music revenues worldwide
- 250+ m registered ebay users, 50+ k product categories
- 200+ m streams from YouTube a day
- 200+ m myspace users
- Over 7 m users in Second Life
- 100+ bn emails per day, incl. 45 bn spam
- Over 100 m blogs worldwide
- 3.0+ bn mobile phone users
- 3+ bn SMS per day
- 1+ bn mobile phones with camera
- 3+ bn € mobile gaming revenue in
- 8+ m mobile TV users in Japan and South Korea

1 Trend to Digitize — *ALL IP / ALL Digital*

2 Trend to Socialize — *Global Communications & Local Actions*

3 Trend to Individualize — *My Tech NOT Hi Tech choice & the "long-tail"*

Source: BT

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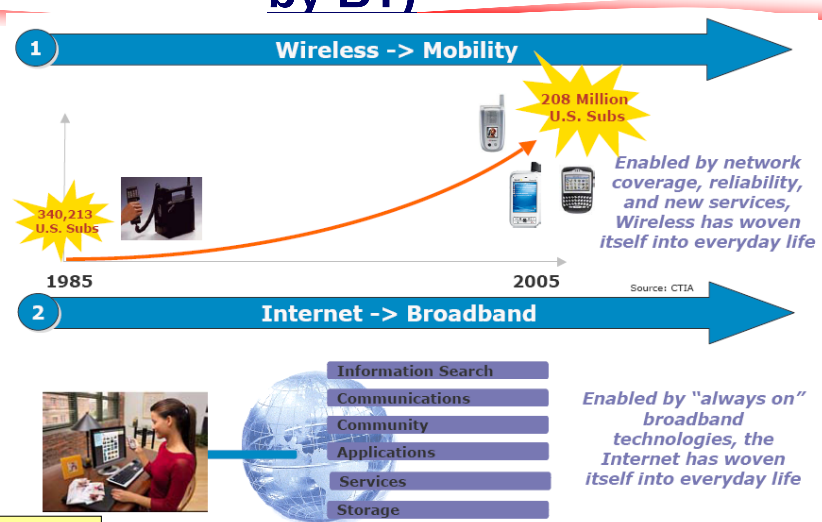
Dynamics and Trends (as seen by BT)



Source: BT

Mega Trends of Mobility & Internet Have Become Pervasive

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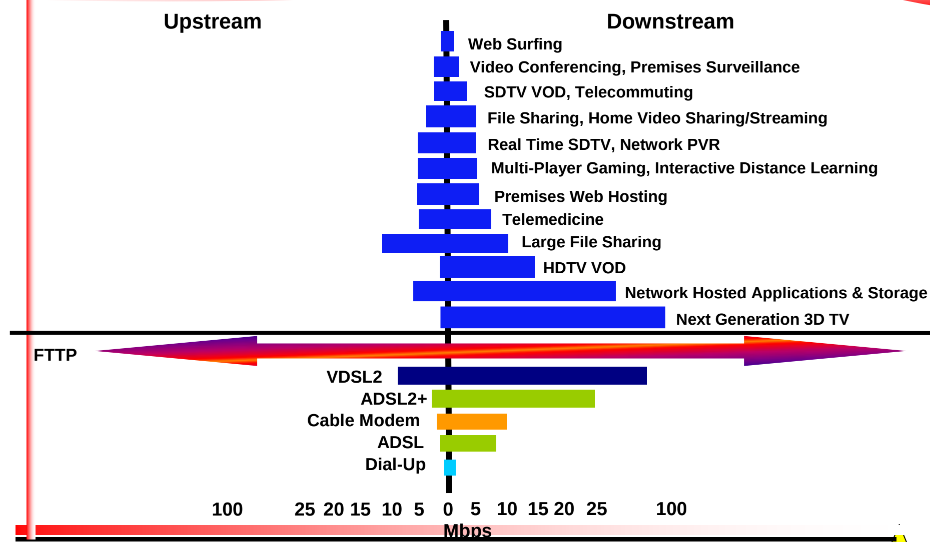
Applications and Media Bandwidth



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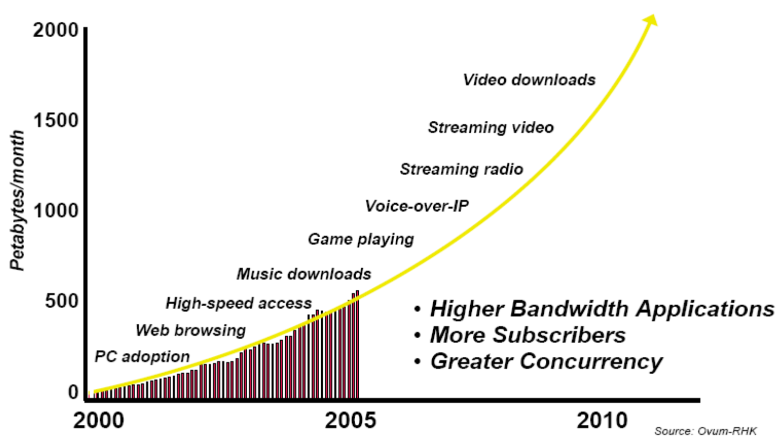
Traffic Growth Prediction



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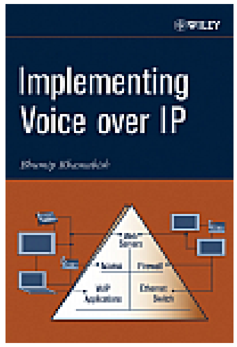
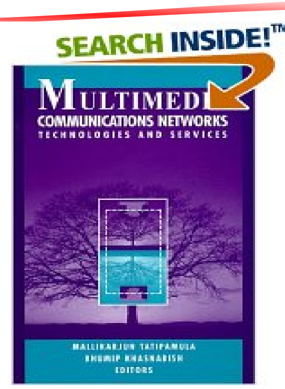
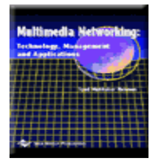
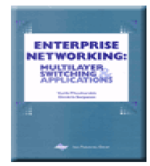
A Few Useful Books









[1] Chapter 2 & Appendix-C of "*Implementing Voice over IP*," by Bhumip Khasnabish, Published by Wiley-IEEE, 2003, ISBN 0471216666, 9780471216667, 208 pages.

[2] Chapter 3, 4, and 6 of "*Multimedia Communications Networks: Technologies and Services*," Edited by by Mallikarjun Tatipamula, and Bhumip Khasnabish, Artech House, 1998, ISBN 0890069360, 9780890069363, 631 pages.

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Thanks for Your Attention and Participation!

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