Converged Services and a New Generation of Networking

... your Comments please! ...

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Overall Presentation Outline

- Convergence of Communications
  - VoIP, IPTV, Streaming media
  - TeleMediCare and TelePresence/TruePresence
- Architecture for New Generation of Networking
- Research Topics, and Emerging Revenue Models
- Q&A and Open Discussions
- Extras: based on time availability & interests
  - Wireline and Wireless Broadband Access
  - Multimedia Traffic Transmission Techniques

Outline of this Section

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

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

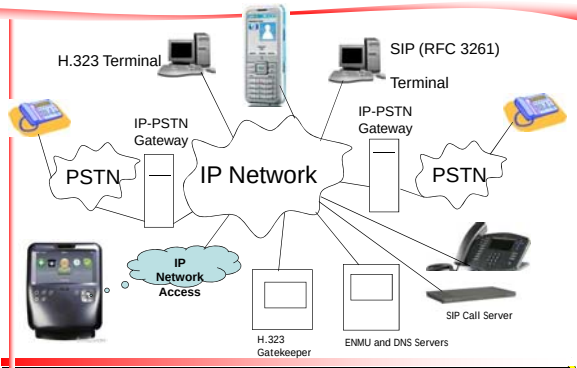
Today's Communications Services

- 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

... Voice Service Evolution ...

Voice over IP Network Elements

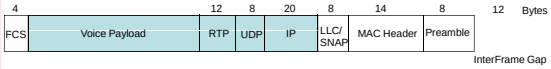


VoIP Protocol Stack

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					

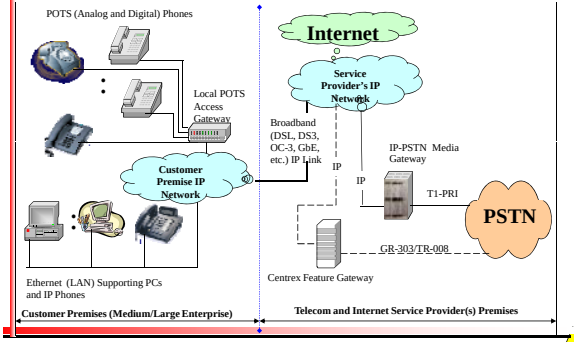
Bandwidth Estimate for Voice over IP/Ethernet

- 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)

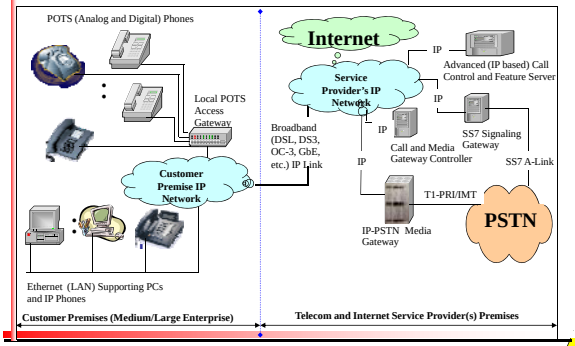


$$\text{Bandwidth} = \frac{(4+ \text{Voice Payload} + 40 + 8 + 14 + 8 + 12) \times 8}{\text{Packetization Delay}}$$

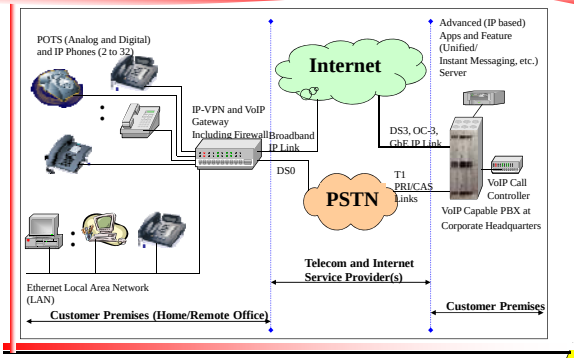
VoIP Based Centrex Service (legacy)



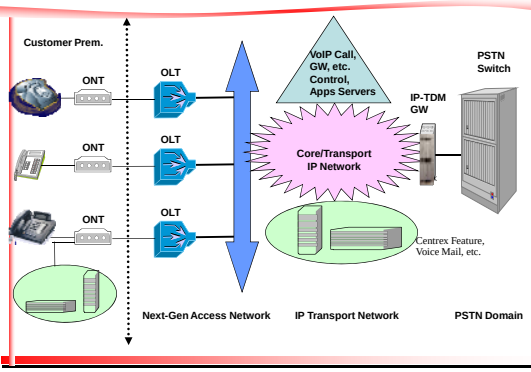
VoIP Based Centrex Service (Emerging)



VoIP for Remote/Traveling Tele-workers



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

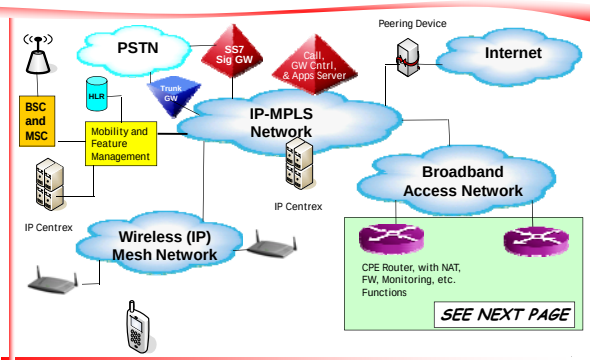
VoIP over Wireless Access (../1)

- 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.

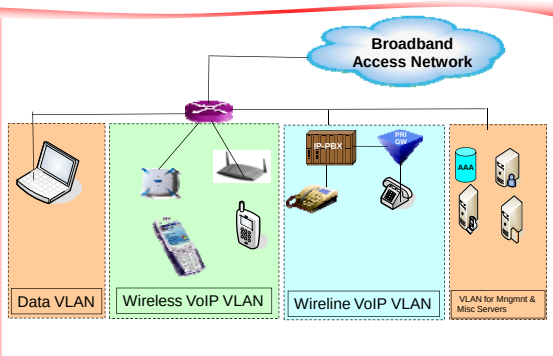
VoIP over Wireless Access (../2)

- VoIP over Wireless phones
  - Dual (SIP and other VoIP clients over WiFi, xCDMA, GS, LTE, WiMax) 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, LTE, WiMax, … …)
    - 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)



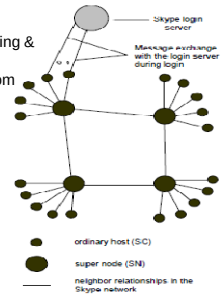
VoIP over Wireless Access (../4)



Wireshark Capture of SIP Messages (partial)

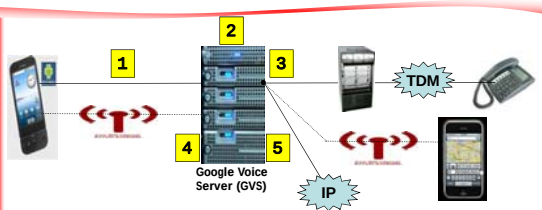
Skype Voice Service

- 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

Google Voice Service



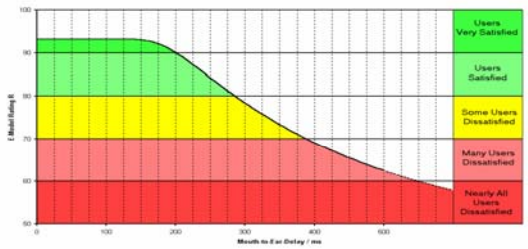
1-Call Originates from a Registered Endpoint
2-GVS Accepts the Call
3-GVS Forks the Call
4-Subscriber Accepts the Call via One Endpoint/Device
5-GVS Bridges the Call Legs & Activates Subscribed Features

http://www.youtube.com/results?search_query=google+voice

Google Voice/Talk Service

- Google registered a million or so phone numbers in preparations to launch Google Voice beyond a private beta. Google plans to support 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

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.

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

Skype goes Mobile!

- Verizon Wireless made **Skype mobile** an easy-to-download and free app for **3G Smartphone** customers
- **Skype-to-Skype calls and chats are unlimited and free** when initiated with Skype mobile from a Verizon Wireless 3G Smartphone with a data plan (Skype has more than 580 million worldwide users)
- Skype mobile **calls made to domestic landline** and wireless numbers **use minutes from customers' voice plans**, and Skype mobile calls to international numbers are charged Skype's low, international rates.

Source: http://www.verizonwireless.com/skypemobile/

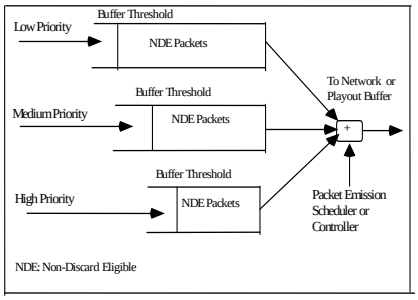
Factors Affecting the Quality of Experience of a Service

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

Nodal QoS via Packet Prioritization

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

Nodal QoS via Packet Prioritization (…/2)



Nodal QoS via Packet Prioritization (…/3)

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

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

$$\gamma = \frac{2x(0.95-1.0)}{(0.95\,x\,3.24)+0.6} = \frac{-0.10}{3.678} = -0.02719$$

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

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

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

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

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%)

$$BHCA = \frac{[Network.Calls.per.Sec x 3600]}{Total.NO.of.Subscribers}$$

$$Offered.Load(Erlang) = \frac{Total.No.of.Subscribers x (BHCA x AHT)}{3600}$$

$$NO.of.Ports.Req. = [Offered.Load(Erlang)] x Blocking.Factor$$

ETE QoS and SLA

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

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

- 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.

Costs for QoS and SLA

$$Profit = [Revenue - AllCosts]$$

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

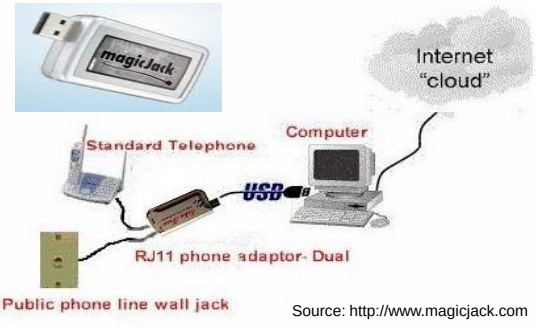
$$Profit Margin(percentage) = \frac{Profit}{Total\ Revenue}$$

Emerging Communications Services

- High-Definition, Stereo, Surround Sound Voice
- 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)

An Emerging version of the OTT Voice Service



Source: http://www.magicjack.com

Cloud-based and Virtualized PBX

- Open VoIP Peering
  - Target is to use **Standard SIP protocol** instead of native proprietary VoIP protocol, server (H-PBX or Hosted PBX) endpoints
  - **Robustness** and **features** of traditional H-PBX along with **cost-effectiveness of VoIP**
  - Low cost **Standard implementation** of virtual receptionist, ACD queuing, voicemail, conferencing, follow-me forwarding, and more

Devices for Converged Services

- *Device is "becoming" the service* in the emerging information-distribution, communications, and entertainment environment
  - This essentially calls for the support of **converged services using any device** at hand by embedding the clients and capabilities as the basis of demands and requirements
- The Network must inherently support **cognitive access** and desired QoS/QoE for all of the required services – voice, data, video, graphics, gaming, and so on
  - **Smart** and **intelligent** devices will use the **network as transport commodity;** just like the cars use the city roads and highways (?? !!)
- Dynamic groups of user will be formed just like the **cars can form a cluster** based on the proximity for video games, collaboration, etc.

2009 US Wireless Stats

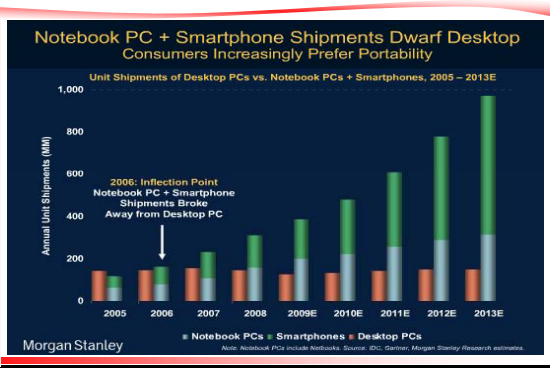
(Subscribers and Retail Stores)

Source: WSJ, May 2010

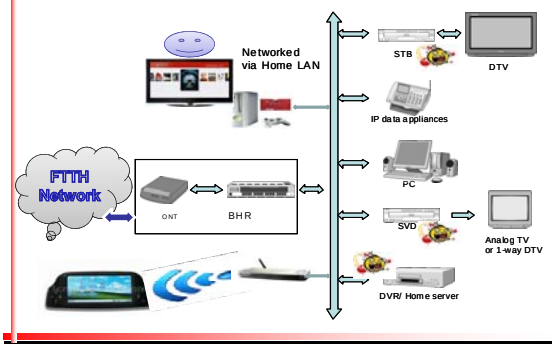
World's Top 8 Mobile Phone Countries

Rank	Country or region	Number of mobile phones	Population	% of population	Last updated
—	World	4,600,000,000	6,797,100,000	60.6	2009[1]
1	China	780,000,000	1,335,330,000	58.5	Mar. 2010[2]
2	India	584,323,402	1,178,071,000	49.6	Mar 2010[3]
3	United States	285,610,580	308,505,000	91.0	Dec. 2009[4] [5]
4	Russia	208,330,000	141,927,297	146.8	Jan. 2010[6] [7]
5	Brazil	180,765,438	191,480,630	94.40	Apr. 2010 [8]
6	Indonesia	140,200,000	231,369,500	60.53	Dec. 2008[9]
7	Japan	107,490,000	127,530,000	84.11	Mar. 2009[10]
8	Germany	107,000,000	81,882,342	130.15	2009[11]

Growth of Smart Devices over Next few Years



Complexity of Home Networks



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 Mallikarjun Tatipamula, and Bhumip Khasnabish, Artech House, 1998, ISBN 0890069360, 9780890069363, 631 pages.

[3] Chapter 4 of "Next Generation Telecommunications Networks, Services, and Management," Edited by T. Plevyak and V. Sahin, Wiley-IEEE, April 2010, ISBN: 978-0-470-57528-4, 297 pages.

IEEE COMMUNICATIONS SOCIETY

Thanks for Your Attention and Participation!

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