CSC 402 – Internet Technology

Recap

- Internet Relay Chat (IRC)
- Broadband Communications and Policy

VoIP

- Voice over IP (VoIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network.
- VoIP can be achieved on any data network that uses IP, like Internet, Intranets and Local Area Networks (LAN).
- Voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network.
- Signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities.
- Main motivations for Internet telephony is the very low cost involved. Some other motivations are:
  - Demand for multimedia communication.
  - Demand for integration of voice and data networks.
- It is also called IP telephony, Internet telephony, voice over broadband, or broadband telephony. Used by Skype, MSN Messenger, Google Talk, etc.

VoIP

- For VoIP to become popular, some key issues need to be resolved:
  - **Quality of voice**: As IP was designed for carrying data, so it does not provide real time guarantees but only provides best effort service. To ensure good quality of voice, we can use either Echo Cancellation, Packet Prioritization (giving higher priority to voice packets) or Forward Error Correction (FEC).
  - **Interoperability**: Products from different vendors need to operate with each other if voice over IP is to become common among users.
  - **Security**: The common tunneling protocol used is Layer 2 Tunneling protocol and the common encryption mechanism used is Secure Sockets Layer (SSL).
  - **Integration with Public Switched Telephone Network(PSTN)**: PSTN and IP telephony network appear as a single network to the users of this service.
  - **Scalability**: VoIP systems needs to be flexible enough to grow to large user market and allow a mix of private and public services.

VoIP Protocols

- RTP – Real-time Transport Protocol (BUT built on top of the UDP and TCP -> application-layer protocol).
  - Defines a standardized packet format for delivering audio and video over the Internet.
  - Session initiation must be performed by other protocol – typically SIP.
- RTP can carry any data with real-time characteristics.
- Out of order delivery is still possible, no support for flow and congestion control.
  - Data necessary for putting the received packets in the correct order is delivered.
- RTCP (Real Time Control Protocol) provides information about reception quality which can be used to make local adjustments.
  - In case of congestion, the application could decide to lower the data rate.

VoIP Protocols

- The commercial VoIP software was introduced in 1995 by Vocaltec.
  - Designed for home PC.
  - Uses H.323 Protocol.
- H.232 is ITU-T's (International Telecommunications Union) standard that vendors should comply while providing Voice over IP service.
- Components of H.232:
  - **Terminals**: LAN client endpoints that provide real time, two way communications.
  - **Gateways**: Endpoint on the network which provides for real-time, two-way communications between H.323 terminals on the IP network and other ITU terminals on a switched based network, or to another H.323 gateway.
    - They are "translator" i.e. they translate different transmission formats, e.g from H.225 to H.221.
    - They are also capable of translating between audio and video codecs and act as the interface between the PSTN and the Internet.
    - They take voice from circuit switched PSTN and place it on the public Internet and vice versa.
    - In a single LAN gateways are optional, when the terminals on a network need to communicate with an endpoint in some other network, then they communicate via gateways using the H.245 and Q.931 protocols.

VoIP Protocols

- Components of H.232:
  - **Gatekeepers**: Most vital component of the H.323 system and dispatches the duties of a "manager". It acts as the central point for all calls within its zone (A zone is the aggregation of the gatekeeper and the endpoints registered with it) and provides services to the registered endpoints.
    - Functionalities include address translation, admission control, call signaling, call authorization, bandwidth management, and call management.
  - **Multipoint Control Unit (MCU)**: The MCU is an endpoint on the network that provides the capability for three or more terminals and gateways to participate in a multipoint conference.
- VoIP is used predominantly for personal level use rather than corporate or enterprise level.

VoIP Protocols

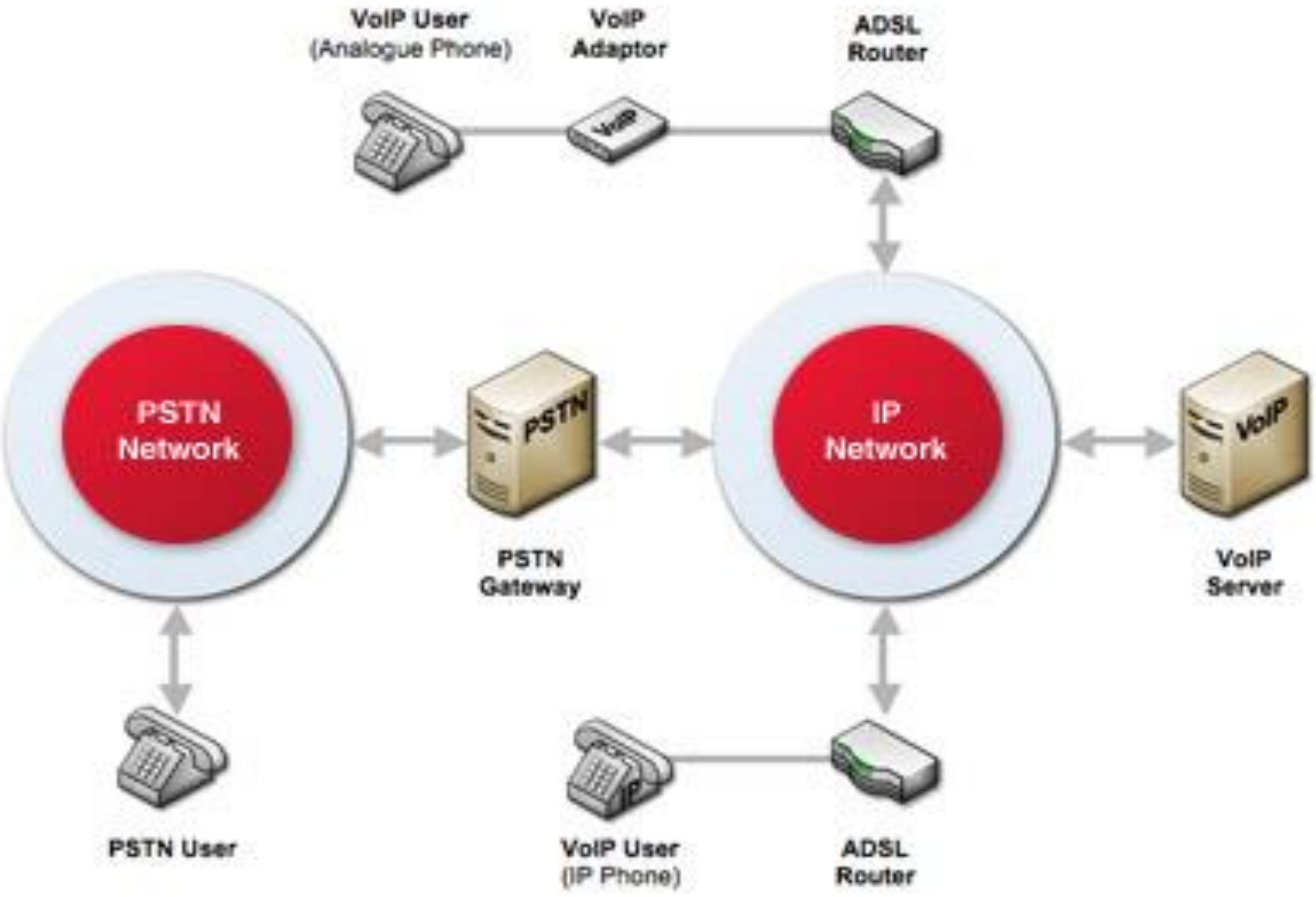
- Session Initiation Protocol (SIP) – application-layer control (signaling) protocol for creating, modifying, and terminating sessions.
  - Does not depend on the transport protocol – works with UDP and TCP.
  - Does not depend on the type of session to be established.
  - Simple and lightweight.
  - Text-based, allowing for low overhead.
- SIP is primarily used in setting up and tearing down voice or video calls. All voice/video transmission is done over separate session protocols, typically RTP.
- SIP can mediate any kind of communication session from voice to video or future, yet unrealized applications.
- SIP Services includes:
  - **User Location**: determination of the end system to be used for communication.
  - **Call Setup**: ringing and establishing call parameters at both called and calling party.
  - **User Availability**: determination of the willingness of the called party to engage in communications.
  - **User Capabilities**: determination of the media and media parameters to be used.
  - **Call handling**: the transfer and termination of calls.

VoIP Protocols

H.323	SIP
Complex protocol	Comparatively simpler
Binary representation for its messages	Textual representation
Requires full backward compatibility	Doesn't require full backward compatibility
Not very modular	Very modular
Not very scalable	Highly scalable
Complex signaling	Simple signaling
Large share of market	Backed by IETF
Loop detection is difficult	Loop detection is comparatively easy

VoIP

- How it works:



VoIP Protocols

- Skype uses VoIP protocol.
- Directly competes with SIP and H.323, using proprietary software and proprietary communication protocol.
  - Skype is both the name of the application and the protocol.
  - Overcomes firewall and NAT problems.
  - On top of VoIP provides additional services: instant messaging, file transfer, and video conferencing.
- Almost all the traffic is ciphered.
  - Lack of transparency.

VoIP Protocols

- Skype uses a peer-to-peer network scheme
- Competition often uses client-server scheme
- Data sent on top of TCP and UDP
- Decentralized user directory, distributed among the nodes
- Easily scalable.
  - Currently (Sept 2011) over 660 mln registered users, with about 50 mln users logged in every day.
  - The number of simultanious online users keeps growing as Skype becomes popular on mobile phones.
    - January 2011 - 27 million users.
    - February 2012 – over 32 million.
    - 05 March 2012 – broken 35 million mark

VoIP Protocols

- Skype Security:
- Skype uses extreme countermeasures against reverse engineering of its software and protocol.
  - Anti debugging techniques.
  - Use of obfuscated (disguised) code.
  - It uses network bandwidth, even when idle.
  - China casus (TOM-Skype SW – Internet censorship).
- Login process is secured using a combination of RSA, AES and MD5.
- An disguised list of login servers is hardcoded in the Skype exec file.
- RC4 is used to cipher the payload of datagrams.

VoIP Protocols

- Random Facts:
- “Sky peer-to-peer” --> “Skyper”.
  - Dropped the ‘r’ because of lack of an available domain name.
- Since it’s release in 2003 it gained over 660 mln users by now.
  - Currently the largest international voice call carrier (by minutes of calls).
- Interesting business ‘problems’ surrounding the Skype purchase by eBay in 2005.
  - eBay bought the Skype company (network) but not the underlying technology.
  - Software licensing dispute.
  - Failed at integrating Skype with their auction platform.
  - Total valuation dropped from $4.1 bln to $2.75 bln.
  - Finally eBay sold back most of the shares in 2009.
- In May 2011 Microsoft has bought Skype for $8.5 bln.

VoIP

- Advantages:
  - Cheaper call rates.
  - Calling person need not necessary to receive call.
  - Better Voice Quality Using Wideband Codecs.
  - Adding new features and applications over time is easy.
  - Integration of voice, data, fax, video is possible.
  - Features
    - FoIP
    - Voicemail
  - Automated calls (alarms, evac systems).

- Limitations:
  - Packet Delay.
  - Packet Loss (no guarantee of delivering packets) i.e. couldn’t hear properly at the other end or pixelated video.
  - Jitter (variable delay).
  - Traffic Sharing.
  - Spending the whole call saying ‘can you hear me now’.
  - People are very intolerant of voice dropouts on ‘normal’ phones.

FoIP

- Fax over Internet Protocol.
- FoIP has lagged behind VoIP because fax communications are rarely the dominant form of telephony communication for a business.
- Migrating an organization’s main form of telephony communication over to IP was the first priority and this is almost always voice traffic.
- As VoIP has matured, organizations continue to push towards a comprehensive Unified Communications solution where IP is the backbone for all communications, including fax.

FoIP – Fax Specifications

Group Designation	Relevant ITUT Specifications	Transmission Time
G1 (Group 1)	T.2	6 minutes
G2 (Group 2)	T.3	3 minutes
G3 (Group 3)	T.30, T.4, and T.6	1 minute or less
Super Group 3 (SG3)	T.30 and T.6	Less than a minute
G4 (Group 4)	T.6, T.503, T.521, T.563, T.72, T.62, T.62 bis, T.70, and F.161	Less than a minute

FoIP

- The two main classifications of fax communications encountered today
  - Group 3 (G3)
  - Super Group 3 (SG3)
- The older G1 and G2 fax groupings were replaced by G3 and are no longer used.
  - The G4 standard has never been widely implemented.
- Therefore, the G3 and SG3 classifications are the only ones that are relevant when dealing with FoIP.
- T.30 protocol handles call setup and other negotiations for the sending of fax page information.
  - It also handles parameters such as paper size, image resolution, fax page encoding algorithm, and page transmission speed are exchanged in T.30 messages between the fax endpoints.
- T.30 messages are sent at a slow speed of only 300 bps, much higher speeds are negotiated for the sending of the actual fax pages.
- T.4 and T.6 define encoding or compression algorithm utilized for the fax page transmission.

FoIP

- FoIP transport methods:
  - Pass through.
  - Relay.
  - T.37 Store-and-Forward fax.
- IP interconnection: IP to act as a ship that carries different technologies from source to destination.

Unified Messaging System (UM)

- Integration of different electronic messaging & communications media (e-mail, SMS, Fax, voicemail, video messaging, etc.) technologies into a single interface, accessible from a variety of different devices.
- UM solutions integrate communications processes into the existing IT infrastructure, i. e. into CRM, ERP & mail systems.
- Handling of voice, fax, and regular text messages as objects in a single mailbox that a user can access either with a regular e-mail client or by telephone.
- Cisco Unity Unified Messaging - Access Multiple Message Types from One Inbox