Computer Networks

Topic 11:

Voice over IP and Video Conferencing

Computer Networks

Topic 11 – Lecture 1:

Voice over IP (VoIP)

Scope and Coverage

This topic will cover:

- Voice over IP (VoIP)
- Installing and configuring voice networks and video conferencing

Learning Outcomes

By the end of this topic, students will be able to:

- Describe the components of a VoIP system
- Install and configure a VoIP system
- Install and configure a web-based video conferencing solution

Communication Over the Internet

- Typical transmission over IP in the early days
  - Data
  - Simple web pages
- Media transmission over IP today
  - Audio
    - Telephony (VoIP), live radio, etc.
  - Image
    - Gallery viewers, CAD drawings, etc.
  - Video
    - Streaming video, “live” TV, etc.

What is VoIP?

- The use of IP networks, namely the LAN (internal network) and WAN (the Internet) to carry voice communications
- Voice was originally carried over circuit switched networks (and still is)
  - PSTN
- Internet that was originally designed to carry computer data is also used to carry voice.
  - A packet switched network

Basic VoIP Functions

- Signalling
- Database services
  - Mapping addresses (IP to Phone numbers) etc.
- Call connect and disconnect (bearer control)
- Codec operations
  - For encapsulating voice signals into data packets.
  - For decoding the packets and transforming them back into a voice signal.

What is a Codec?

- A device or software that is used to compress or decompress a digital media file, such as an audio file
- A codec can consist of two components:
  - An ***encoder*** performs the compression (encoding) function.
  - A ***decoder*** performs the decompression (decoding) function.

VoIP Components

- Servers
  - For processing calls and managing interactions with the telephone network, etc.
- End-point devices, such as telephone handsets
- Media and VoIP gateways
- IP network

VoIP Gateways & Gatekeepers

- Gateway equipment performs the task of allowing non-IP equipment to talk to IP equipment.

- Gatekeepers manage the calls within a particular zone.

Issues

- Latency
- Jitter
- Bandwidth
- Packet loss
- Reliability
- Scalability
- Security
- Features
- Interoperability
- Switch over cost

Latency

- The time taken for a packet to arrive at its destination
- This is affected by:
  - Packet switching overhead
  - The amount of traffic on the network
- Latency may result in voice synchronisation problems
  - Gaps in transmission

Jitter

- The delay experienced in receiving a packet when a packet is expected to arrive at a certain time
  - If you send a sequence of packets from point A to point B, each of the packets will need a slightly different time to reach the destination.
  - The varying transit times are not an issue if you are downloading a web page, but they matter if you wish to transmit a stream of real-time data, such as voice, video, etc.

Bandwidth

- Bandwidth is shared between voice and computer data on the network.
- Certain bandwidth may have to be allocated for voice communication on a network.
- What happens when many VoIP calls are occurring at the same time?

Packet Loss

- Packet loss in unavoidable
- Voice transmission can only tolerate minimal packet loss.
  - It should not distort the audio.
- Packet loss on VoIP is similar to when a call on a cell/mobile phone begins to “break-up”.
- This results in the receiver missing parts of the conversation.

Reliability

- The reliability of the network has an impact on the telephony service.
  - In the analogue telephone industry, reliability of 99.999 percent uptime is required.

- Require reliable network connection with sufficient bandwidth.

Scalability

- Ability to add more telephony equipment as the company grows.
- Network bandwidth and other issues may have an effect on scalability.
- Can the network deal with the extra traffic?

Security

- VoIP typically uses the Internet.

- It is, therefore, vulnerable to the same type of security risks as other Internet traffic:
  - Hacking
  - Denial of service
  - Eavesdropping

Features

- In order to compete with traditional telephony, VoIP telephony needs to match and, in the long run, exceed the features provided by the PSTN.
  - Call waiting
  - Conference calls

- Already has extra features:
  - Video calls
  - Free! (in some instances)

Interoperability

- IP telephony equipment manufactured by different vendors must be able to talk to each other.
- Without standardised quality of service mechanisms, businesses would need to buy all the equipment and the QoS server from the same manufacturer.
- VoIP seems to be divided among multiple vendors with reluctance to establish interoperability.

Switch Over Costs

- Switching from a more traditional telephone system to VoIP may involve costs.
  - Equipment
  - Extra bandwidth

- Savings are likely on cost of calls.

Benefits of VoIP

- Issues have been resolved to an acceptable level
- Usually reduces costs
- Utilises “spare” bandwidth
- No separation of data and voice networks
- “Extras” are free of charge:
  - Conference calling
  - Call forwarding
  - Automatic redial
  - Caller ID

Computer Networks

Topic 11 – Lecture 2:

VoIP Systems & Video Conferencing

What is Needed to Set Up VoIP?

- An existing broadband Internet connection
- A VoIP-compatible phone
- A VoIP provider

- Alternatively, use a software solution.
  - E.g. Skype

Broadband Internet Connection

- Essential for telephone-quality audio
- Good connection speed
- If you expect to make a lot of calls, make sure that your plan has a suitable usage limit.
- Check with your ISP – they may also offer a VoIP service and it may be possible to save on both bills by bundling.

VoIP Compatible Phone

- Telephone handsets specifically designed to work over VoIP can be purchased.
- Plug directly into your Internet connection
- ATA adapter can be purchased, which allows plugging of existing telephone into Internet connection.
- Both require a router or network hub - can be wired or wireless
- Some newer routers incorporate VoIP adapters.

VoIP Provider

- Choose a provider and plan that suits your needs:
  - Price
  - Additional features, e.g. multiple numbers for incoming calls
- There are many providers available in many countries.
- Not available in some countries

Software VoIP

- Your computer can be used as a VoIP phone.
- Needs a VoIP software package
  - the most popular of which is Skype
- Allows you to conduct VoIP calls through a microphone and headphones/speakers or a cheap USB phone
- Calling other VoIP software users is usually free and calls to other phones are at low VoIP rates.

VoIP Protocols

- There are both open and proprietary protocols.
- Open protocols are in the public domain.
- Proprietary protocols relate to one vendor.
- Common protocols in VoIP are:
  - H.323
  - Real-time Transport Protocol (RTP)
  - Session Initiation Protocol (SIP)
  - Skype protocol – a proprietary protocol
- Often, combinations of protocols are used.

H.323

- A multimedia conferencing protocol, which includes voice, video, and data conferencing, for use over packet-switched networks
- A framework of how other protocols fit together, including:
  - H.245 - the protocol used to control establishment and closure of media channels within the context of a call.
  - RTP/RTCP - to transmit audio over IP networks.
  - X.691 - the Packed Encoding Rules (PER) used to encode messages for transmission on the network.

Real-Time Transport Protocol

- A standardised packet format for delivering audio and video over IP networks
- Standard defines a pair of protocols:
  - RTP is used for transfer of multimedia data.
  - RTCP is used to periodically send control information and QoS parameters.
- Provides jitter compensation and detection of out of sequence packets
- Supports data transfer to multiple destinations

Session Initiation Protocol

- Very flexible, lightweight protocol for making multimedia calls
- SIP is an end-to-end, client-server, extensible, text-based protocol.
- The design is based upon HTTP and SMTP.
- SIP runs on any transport protocol (TCP, UDP).
- Provides user location and session management
- Does not provide services itself

Skype Protocol

- Skype is a proprietary Internet telephony network based on peer-to-peer architecture, whereas many VoIP solutions are client-server based.
- The protocol's specs are not publicly available.
- The Skype network is not interoperable with most other VoIP networks without proper licensing from Skype.
- Signaling and voice data is encrypted.
- Skype can be used for video calls.

Video Conferencing

- Two or more participants at different sites using computer networks to transmit audio and video data
- Each participant has a video camera, microphone, and speakers mounted on a computer.
- Voices are carried over the network and delivered to the other participant's speakers; whatever images appear in front of the video camera, appear in a window on the other participant's monitor.

Video Conferencing Advantages

- Less expensive than travelling
- Easy access to experts
- Direct access to training, mentoring, etc.
- More personal than a telephone call
- Allows communication on all levels, including body language
- Saves time and money

Components

- Codec
- Video input device
- Video output device
- Audio input device
- Audio output device
- Data transfer network
- Computer

Videoconferencing Systems

- Web camera videoconferencing systems add on to personal computers
  - Connected by VoIP networks
  - Lowest direct cost
  - Quality of service can range from low to very high
- Dedicated videoconferencing systems:
  - Mid range cost
  - Utilise multipoint control units
  - Quality of service can vary from moderate to high

Multipoint Control Units

- A device commonly used to bridge videoconferencing connections
- An endpoint on a LAN that provides the capability for 3 or more terminals and gateways to participate in a multipoint conference
- Consists of:
  - A mandatory Multipoint Controller (MC)
  - Optional Multipoint Processors (MPs)

Telepresence Systems

- Highest capabilities
- Highest cost
- Can involve especially built teleconference rooms with very high levels of audio and video fidelity
- When the proper type and capacity transmission lines are provided between facilities, the quality of service reaches state-of-the-art levels.

References

- Tanenbaum, A.S. & Weatherall, D.J. (2010). *Computer Networks*, 5th edition. Pearson Education.
- Austerberry, D. (2004). *The Technology of Video and Audio Streaming*, 2nd edition. Focal Press.

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Any Questions?