COMP349
Spoken Language Dialogue Systems
Voice over IP

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Today’s Program

• What is Voice over IP?
• What is Internet Telephony?
• Background Information: TCP/IP
• How Does Internet Telephony Work?
• Quality Issues
• Advantages of Internet Telephony
• SIP (Session Initiation Protocol)
What is Voice over IP?

• Voice over IP is a generic term for all types of voice communication using the Internet protocol IP (TCP/IP) instead of traditional circuit switched technology.

• This includes packet technologies used by telecommunication companies to carry voice at the core of their networks in ways that are not controlled by an end user.
What is Internet Telephony?

- Internet telephony is a service that end users decide to use.
- It is a specialised form of Voice over IP.
- Regular voice telephone calls are transmitted via the Internet.
- All or part of the public switched telephone networks is bypassed.
- Internet telephony can occur
  - between computers
  - between a computer and a phone
  - between phones.
Background Information: TCP/IP

- TCP/IP is composed of layers.
- IP - is responsible for moving packets of data from node to node.
- IP forwards each packet based on a four byte destination address (the IP number).
- The Internet authorities assign ranges of numbers to different organizations.
- The organizations assign groups of their numbers to departments.
- The number of unassigned Internet addresses is running out, so a new schema called CIDR (Classless Internet-Domain Routing) is used.
Background Information: TCP/IP

- TCP - is responsible for verifying the correct delivery of data from client to server.
- Data can be lost in the intermediate network.
- TCP adds support to detect errors or lost data and to trigger retransmission until the data is correctly and completely received.
How Does Internet Telephony Work?

• Since all transmissions must be digital, the caller’s voice is digitized.
• This can be done by the telephone company, the Internet service provider or by a personal computer on your desk.
• Next the digital voice is compressed and then separated into packets.
• Using the Internet protocol, the packets
  – are addressed
  – sent across the network
  – reassembled in the proper order at the destination.
How Does Internet Telephony Work?

- During transmission on the Internet, packets
  - may be lost
  - may be delayed
  - may be damaged.
- Conventional error correction would request retransmission.
- For real-time communication this technique would not work.
- Sophisticated error detection and correction algorithms are used.
- They create sound patches to fill the gaps.
How Does Internet Telephony Work?

• Error correction process
  – stores a portion of the incoming speaker’s voice,
  – uses algorithms to guess the content of the missing packets,
  – creates new sound information to enhance the communication.

• After the arrival of the packets
  the transmission is assembled and decomposed
  to restore the data (to an approximation) of the original form.
Quality Issues

• Quality of a voice transmission using packet technology is sometimes inferior to a circuit switched connection.
• The Internet Protocol was not designed for voice.
• Difference of quality is sometimes obvious to a listener.
• As IP technology improves, quality advantage of circuit switched connection will decrease.
Advantages of Internet Telephony

• Telecommunication carriers have already introduced IP into their networks (computer telephony integration).

• Advantages of this technology are:
  – greater efficiency (does not occupy the entire bandwidth)
  – lower costs (30% cheaper for long distance calls)
  – (potentially) higher reliability
  – open architecture (TCP and IP are open standards).

• You don’t need both a phone and a computer.

• You only need a VoIP-enabled computer.
Obtaining a Voice Client (= User Agent)

- GIZMO and X-Lite are two popular SIP phones:
  - GIZMO Project: http://www.gizmoproject.com/
  - CounterPath’s X-Lite: http://counterpath.com/xlitedownload.html
Connecting to Tellme Studio

- Connecting to Tellme Studio is simply a matter of dialing the SIP URL: sip:8005558965@sip.studio.tellme.com
- If everything is working fine, you should hear the greeting: "Tellme Studio. Developer ID?"
- If you have troubles connecting to Tellme Studio, then you are probably sitting behind a firewall.
- SIP uses UDP¹ (and sometimes TCP) on port 5060.
  ¹UDP (= user datagram protocol)
- The voice streams setup by SIP are transported using RTP (another UDP-based protocol).
Interaction with Tellme Studio
About SIP

• SIP (= Session Initiation Protocol)
  – is an application layer-protocol,
  – can establish, manage and terminate voice and video sessions,
  – sessions involve one or more participant,
  – can use unicast or multicast communication,
  – is being developed by the SIP Working Group,
  – has the status of a proposed standard.
SIP Protocol

RTP (Real-Time Transport Protocol provides end-to-end network transport functions)
SIP Protocol

- SIP is a textual protocol based on the client-server model.
- A request invokes a method on a server.
- The request can be sent either over TCP or UDP.
- The most important SIP method is the INVITE method.
- It is used to initiate a call between a client and a server.
- Other methods are ACK, OPTIONS, BYE, CANCEL, and REGISTER.
- SIP uses the Session Description Protocol (SDP) for media description.
How does SIP Work?

- SIP is all about calling people and services.
- SIP supports dialing traditional telephone numbers.
- However SIP’s native concept of an address is a SIP URL.
- If you want to call Joe, you don’t really want to care which "phone“ Joe uses to answer the call.
- The routing that the call makes depend on who is trying to call.
- If Joe’s wife calls him while he is out at lunch, he might want the call be routed to his mobile phone.
- If his boss makes a call at lunch time, it get routed to his voice-mail.
SIP Session

- SIP sessions utilize up to four major components:
  - SIP user agents,
  - SIP registrar servers,
  - SIP proxy servers,
  - SIP redirect servers.
SIP User Agents (UAs)

- UAs are end-user devices:
  - mobile phone
  - multimedia handsets
  - personal computers, etc.
- The UA client initiates the message.
- The UA server responds to it.
SIP Registrar Servers

- SIP registrar servers are databases.
- They contain the location of all user agents within a domain.
- In SIP messaging, these servers
  - retrieve and send participant’s IP address and
  - other pertinent information to the SIP proxy server.
SIP Proxy Servers

- SIP proxy servers
  - accept session requests made by a SIP UA,
  - query the SIP registrar server to obtain the recipient UA’s addressing information,
  - forward the session invitation
    - to the recipient UA if it is located in the same domain or
    - to a proxy server if the UA resides in another domain.
SIP Redirect Servers

- SIP redirect servers allow SIP proxy server to direct SIP session invitations to external domains.
- SIP redirect servers may reside in the same hardware as SIP registrar server and SIP proxy servers.
Establishing a SIP Session (within the same domain)

- Let’s assume two users who subscribe to the same ISP.
- Hence, they use the same domain.
- User A relies on a SIP phone.
- User B has a PC running a soft client (voice + video).
- Upon powering up, both users register their availability and their IP address with the SIP proxy server (SPS) in the ISP’s network.
- User A tells the SPS that he wants to contact user B.
Establishing a SIP Session (within the same domain)

• The SPS then asks for and receives user B’s IP address from the SIP registrar server.
• The SPS relays user A’s invitation to communicate with user B, including the medium or media user A want to use (via SDP).
• User B informs the SPS that user A’s invitation is acceptable and that s/he is ready to receive the message.
• The SPS communicates this to user A, establishing the SIP session.
• The user agents then create a point-to-point RTP connection enabling them to interact.
Illustration (within the same domain)

1. Call User B
2. Query “Where is User B?”
3. Response “User B SIP Address”
4. ‘Proxied’ Call
5. Response
6. Response
7. Multimodal Chanel Established

Non-SIP Queried (i.e. Database Lookup)
SIP Signaling
RTP
Establishing a SIP Session (across different domains)

• Let’s assume that the user B uses a multimedia handset and is outside of user A’s domain.

• When the user A invites the user B for a SIP session, the SPS in domain A recognises that user B is outside its domain.

• The SPS queries the SIP redirect server for user B’s IP address.

• The SIP redirect server can reside in domain A or B.

• The SIP redirect server feeds user B’s contact information back to the SPS.
Establishing a SIP Session (across different domains)

- The SPS forwards the SIP session invitation to the SPS in domain B.
- The domain B SPS delivers user A’s invitation to user B.
- User B forwards his/her acceptance along the same path the invitation travelled.
Illustration (different domains)

1. Call User B
2. Query “How do I get to User B, Domain B?”
3. Response “Address of Proxy Controller for Domain”
4. Call ‘Proxied’ to SIP Proxy for Domain B
5. Query “Where is User B?”
6. User B’s Address
7. Proxied Call
8. Response
9. Response
10. Response
11. Multimedia Channel Established
Skype

- Skype is a peer-to-peer internet telephony network.
- Skype uses a proprietary protocol and not SIP.
- Skype users can
  - speak to other Skype users for free,
  - call traditional telephone numbers for a fee,
  - receive calls from traditional phones for a fee,
  - receive voicemail messages for a fee.
- Skype conference calls can host up to 10 users.
Skype Usage and Traffic

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Skype

On August 16, 2007, Skype’s peer-to-peer network collapsed for the first time during its four year history. It was down for almost 48 hours. It is believed that the outage was caused by millions of users simultaneously rebooting their machines following a minor Microsoft security update.[3][4]

• Source: http://en.wikipedia.org/wiki/Skype
Take-Home Messages

• Voice over IP uses the Internet instead of PSTN.
• Regular voice telephone calls can be transmitted via the Internet.
• Error correction algorithms are used to create sound to fill the gap if packets are lost or damaged.
• SIP is a session initiation protocol and resembles HTTP.
• SIP provides the capability to call a party at a local-independent address.