VoIP – telephony over internet

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Internet is Packet Switched

- Information – transmitted in small chunks
- Header added to each chunk
- Header at least consists of (usually)
  - Source address
  - Destination address
  - Each intermediate nodes where the packet to be sent
  - Destination can find, who sent the packet.
- Trailer – usually cheksum for error detection

- Creating small chunk means
  - Header and trailer – much larger.
  - Overhead (extra bytes sent per information byte) is large – inefficient system.
- Creating large chunk means
  - More time to create sufficient data for large chunk.
  - 10000 byte chunk from voice – need 1000*125μs = 1.25s
- Usually moderate size packets sent for all realtime traffic.
- Large size packets for ftp/http kind of transactions.
- Switching of information packets from source to destination
  - done by routers in the packets switched network.
  - Only on the basis of headers of the packets.

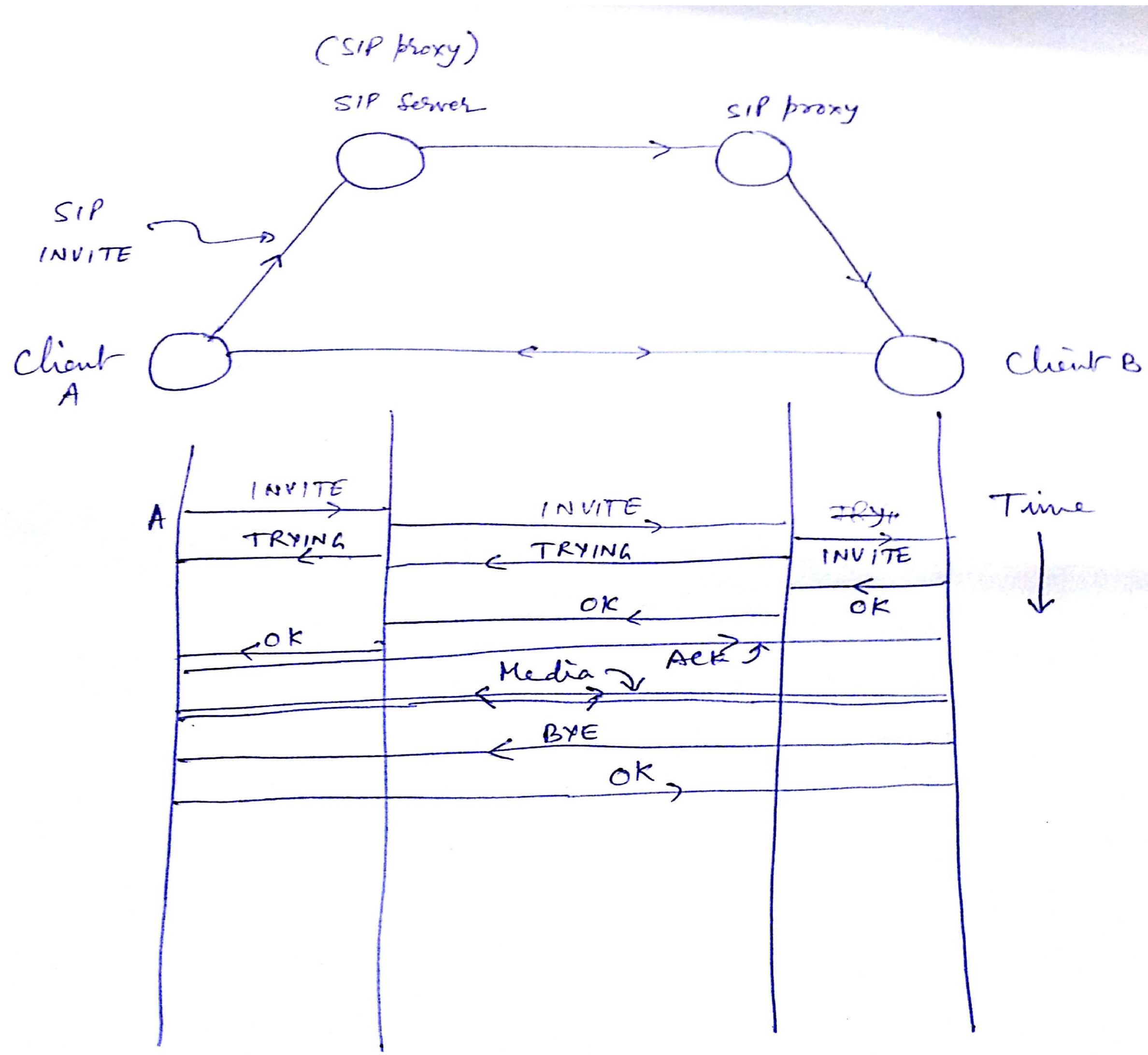
Voice over packet switched network

- Commonly known as VoIP (voice over internet protocol)
- TCP/IP – prominent packet switched network
- For VoIP call, source should know the identity of the destination
  - Identity of application which generate the audio for listener.
- Identity of the destination software process – IP address (32 bits), port number (16 bits)

Indexing server

- Usually hosts on IP network, assigned IP addresses dyanmically
- IP address cannot be used as identity.
- Permanent identity to IP address mapping needed.
- Maintained at indexing server.

- Each client contacts indexing server (also called call-manager)
- Authenticates itself (phone number, password)
- Registers its IP address
- Call-manager and Sip-registrar both are part of sip server.


(SIP proxy)
SIP Server
SIP proxy
SIP
INVITE
Client
A
Client B
A
INVITE
TRYING
INVITE
TRYING
OK
INVITE
OK
OK
ACK
Media
BYE
OK
Time

call setup?

- Caller send a call-setup request to its call-manager (INVITE)
- The request contains the destination (callee) id and its domain name – sip(s) uri
- In the existing situation, the destination domain (telecom operator) identified by first few digits of the phone number.
- Call-manager for different domains – supposed to know each other.
- If not, the call manager will transfer the messages via other known call managers

- Once the call-manager of callee get the request message.
- It looks into the database managed by sip registrar
- Finds the current IP address and port of the callee client.
- Can be multiple clients attached to same sip identity.
- All can be signalled simultaneously about incoming call request.
- One after another after timeouts.
- Registrar can also provide for redirection sip uri.

- INVITE contains the SDP message
- SDP – session description protocol – description of media types, codecs, ports, transport (TCP/UDP), encryption etc.
- The response from callee instrument to its sip server.
- Contain SDP message of agreed upon media descriptions.
- The response flows back via specified intermediate sip servers.

- The calling instrument on receipt of response
  - Sets up the media using agreed upon description.
- Calling and callee party now interacts.
- The media does not flow through SIP server, it is now flowing as IP packets between endpoints directly.

Media gateways

- Sometimes, callee and calling parties informed of media gateway as the other end point.
- SIP server directly controls the media gateway to interconnect the callee and calling party.
- Call can be tapped using media gateway.
- Old telephony systems can be connected to media gateway – internet side sees the convention telephone as VoIP phone, conventional side sees VoIP phone as conventional phone.
- Media gateway does translation between conventional circuit switched telephony, SS7 signalling to packet switched VoIP and SIP signalling.