COMPSCI 314 S1 C

Streaming Audio & Video,

VoIP, Internet Telephony

Not in 2006 exam!

Audio Compression & Encoding

- A codec encodes audio/video/etc. for transmission, and decodes it upon reception. Each codec's behaviour is defined by a standard
- Audio codecs
  - Use various physical models *[Halsall A.2.4]*
  - Trade *bandwidth* against *quality* (measured using MOS)
  - G.721, G.722: ITU-T standards, used in telephone networks
  - MPEG audio: ISO standard, MP3 "CD quality at 64kb/s per channel"
  - Skype: proprietary, software implementation said to be "very good"
- Standards summary: Halsall Table A.5

Video Compression

- Motion JPEG simply presented a sequence of JPEG frames
  - Easy to edit, not very much compression
- MPEG uses sequences of I, P and B frames to get higher compression
  - I frames carry a whole picture
  - P frames carry forward differences, use those + preceding frame(s) to reconstruct the pictures
  - B frames, differences between preceding and following frame. More work to encode/decode, highest level of compression
- MPEG2 (ISO 13818) for studio-quality audio and video
  - Four levels of video resolution

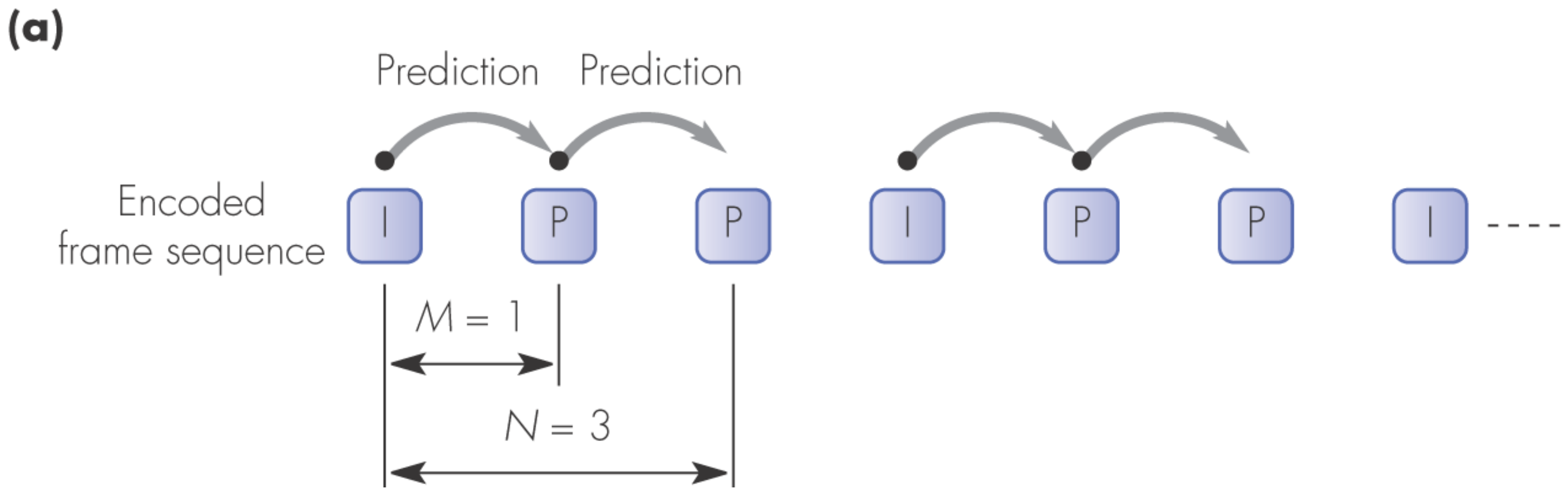

Figure A.17 Example frame sequences with: (a) I- and P-frames only

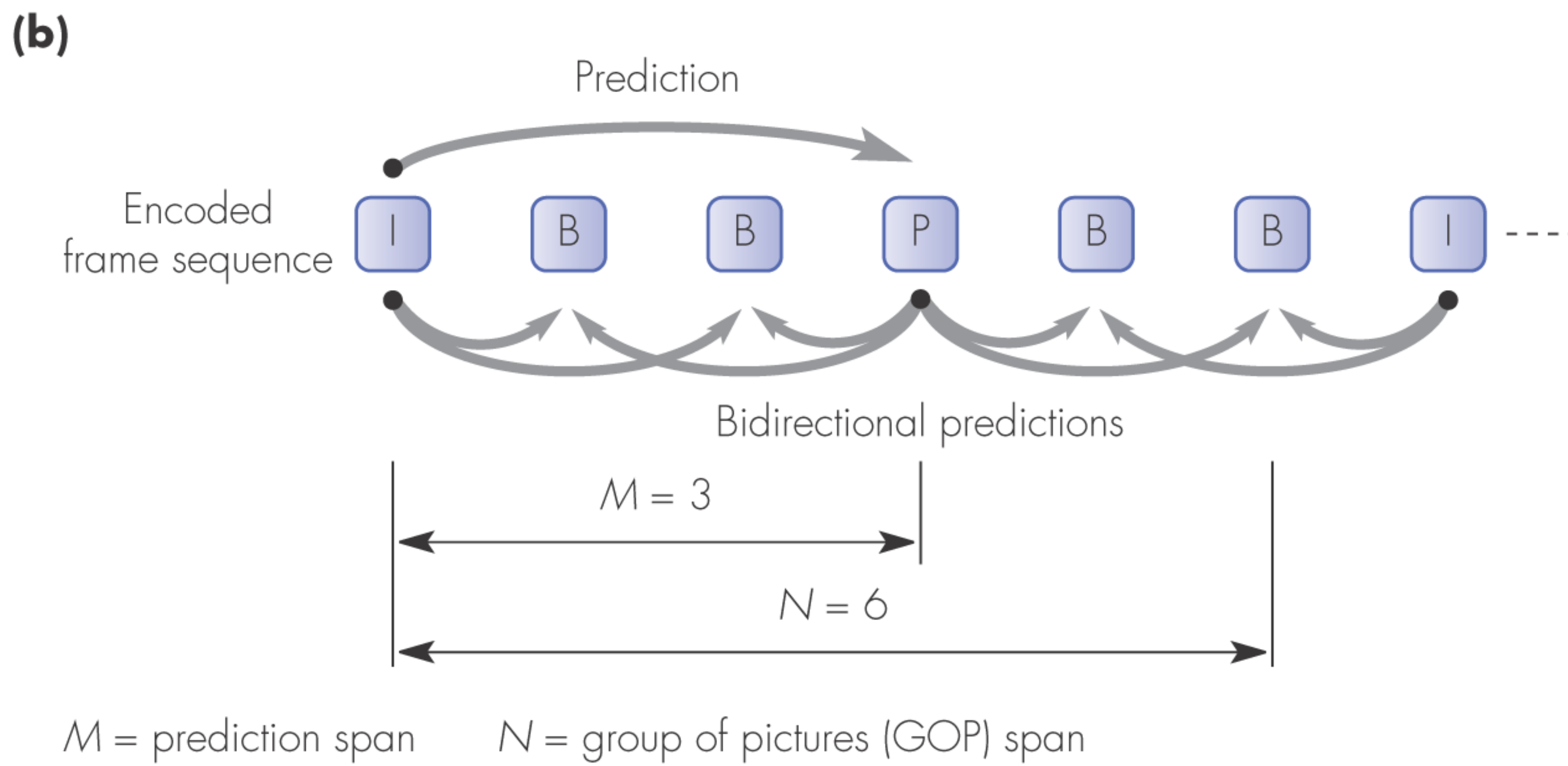

Figure A.17 Example frame sequences with: (b) I-, P- and B-frames

RTP: Realtime Transport Protocol

- Halsall 7.5.1
- An *application* protocol, carried over UDP
- Each packet has a sequence number
  - Used to detect lost and out-of-sequence packets
- And a timestamp
  - Returned to source, used to track RTT
- RTP receiver delivers audio/video via a playout buffer

- Comment about *jitter*: it can mean different things
  - Computer Science: difference in successive transit times
  - Electrical Engineering: differences from expected arrival time
  - Safer to use the term *delay variation*

RTP: Usage Overview

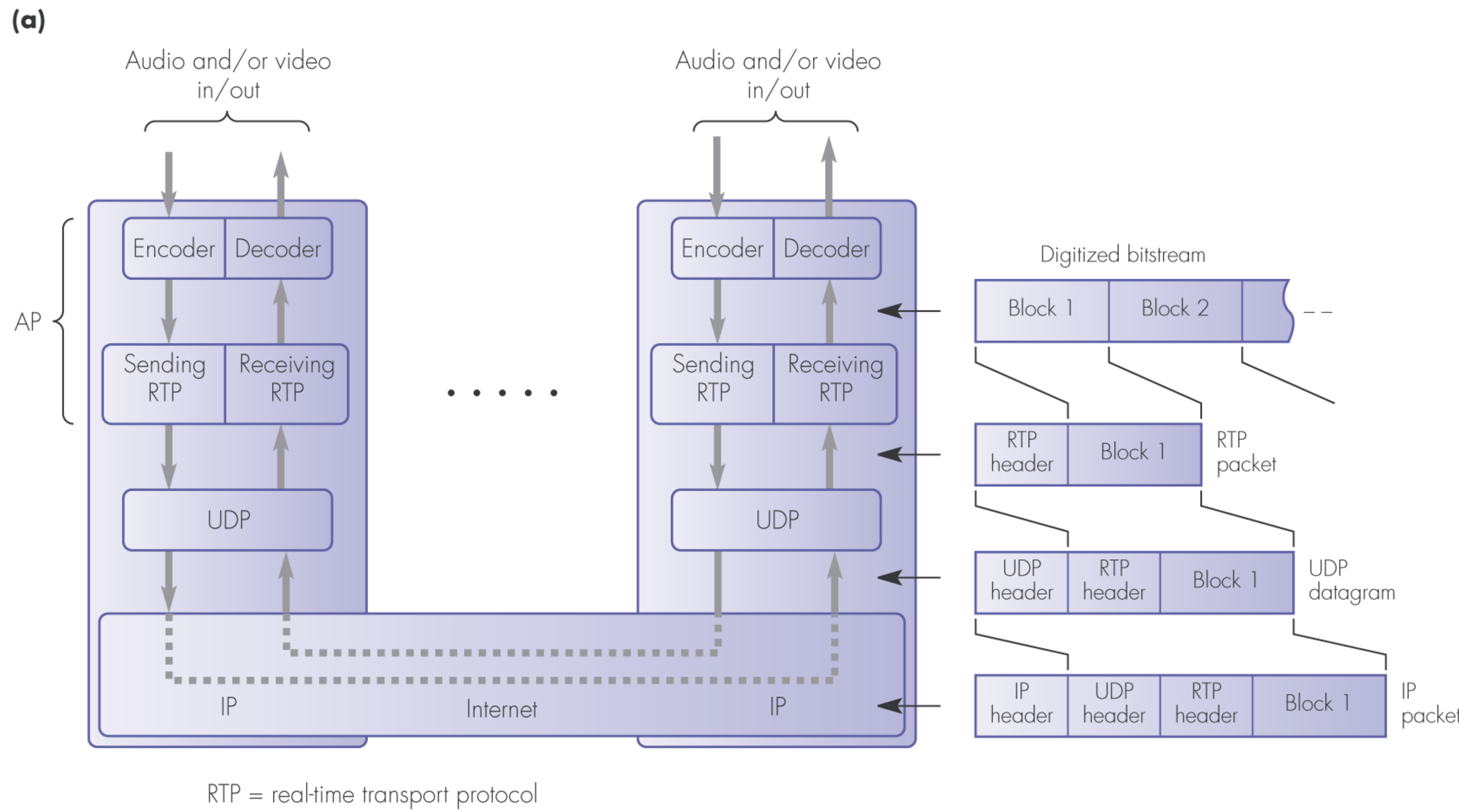

Figure 7.21 Real-time transport protocol: (a) usage

RTP: header format

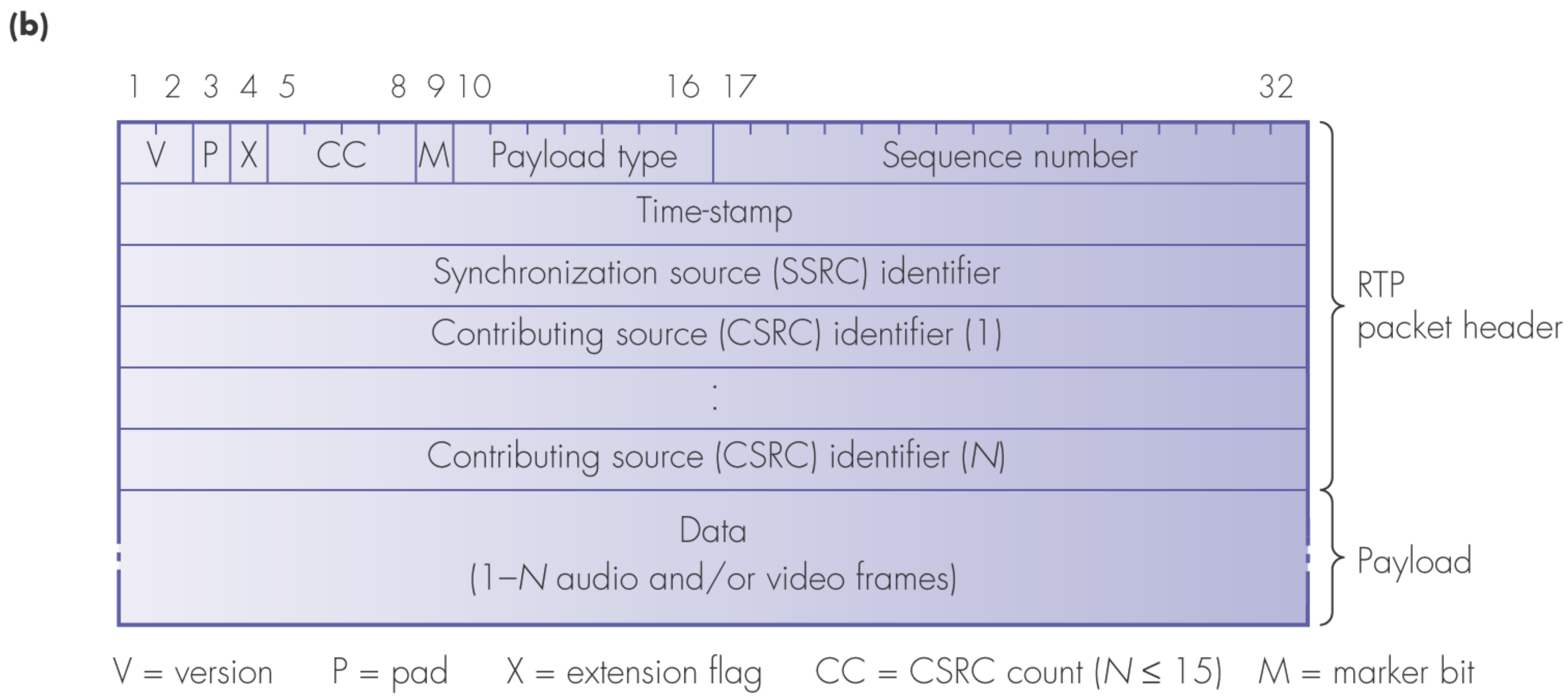

Figure 7.21 Real-time transport protocol: (b) packet format

VoIP & Internet Telephony

- Streamed video
  - QuickTime, RealAudio, Windows Media …
- Early work
  - Desktop audio/video conferencing, CuSeeMe, etc.

- Telephony
  - H.323 (ITU-T)
  - SIP: Session Initiation Protocol
    - User names like email addresses, e.g. *sip:*nevil@auckland.ac.nz
  - E.164 addresses: ITU-T recommendation for PSTN numbering
  - *enum* maps E.164 numbers to domain names, e.g.
    - +64 373 7599 maps to 9.9.5.7.3.7.3.4.6.e164.arpa

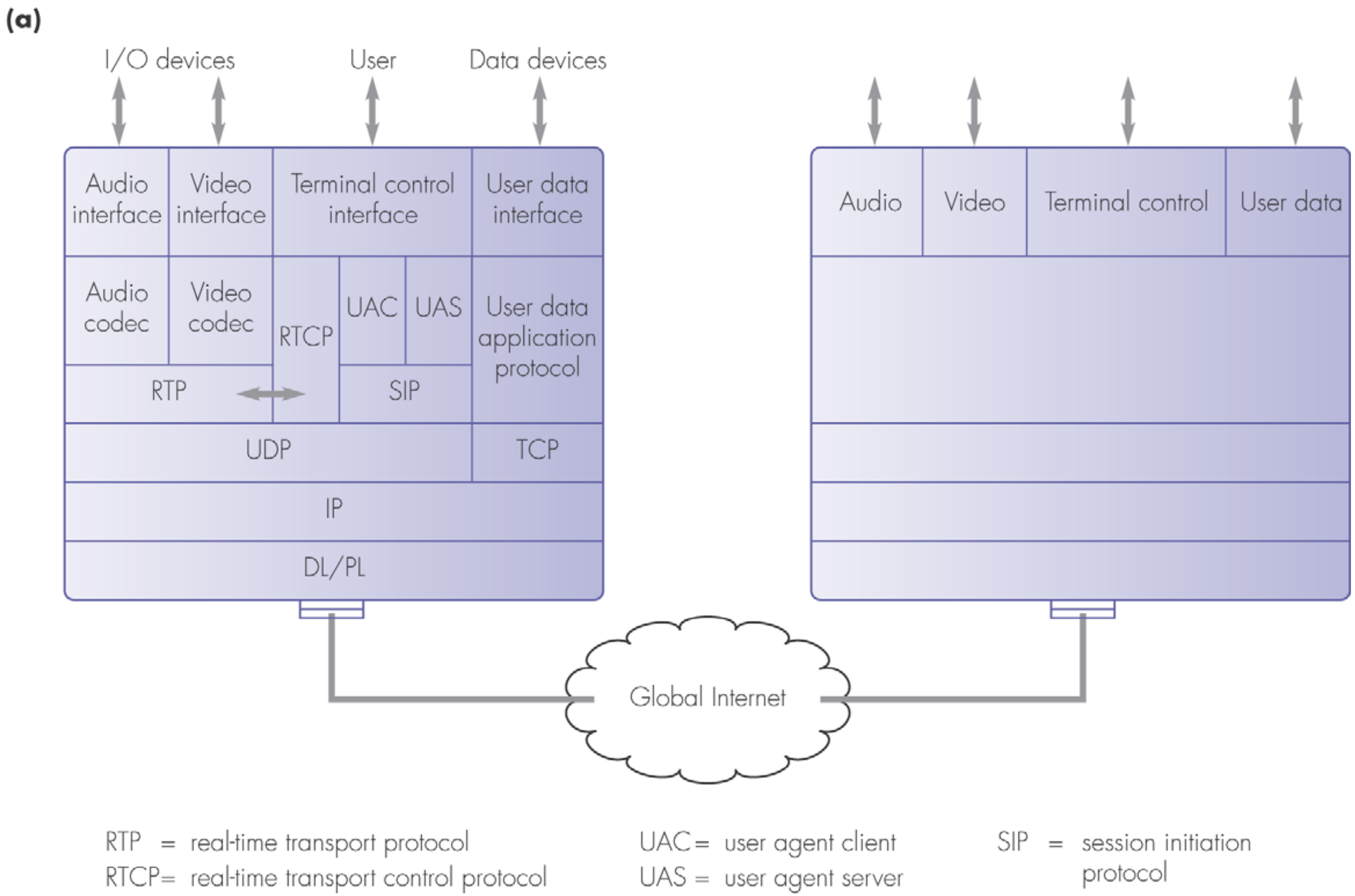

Figure 8.19 Internet telephony: (a) example host device protocol stack

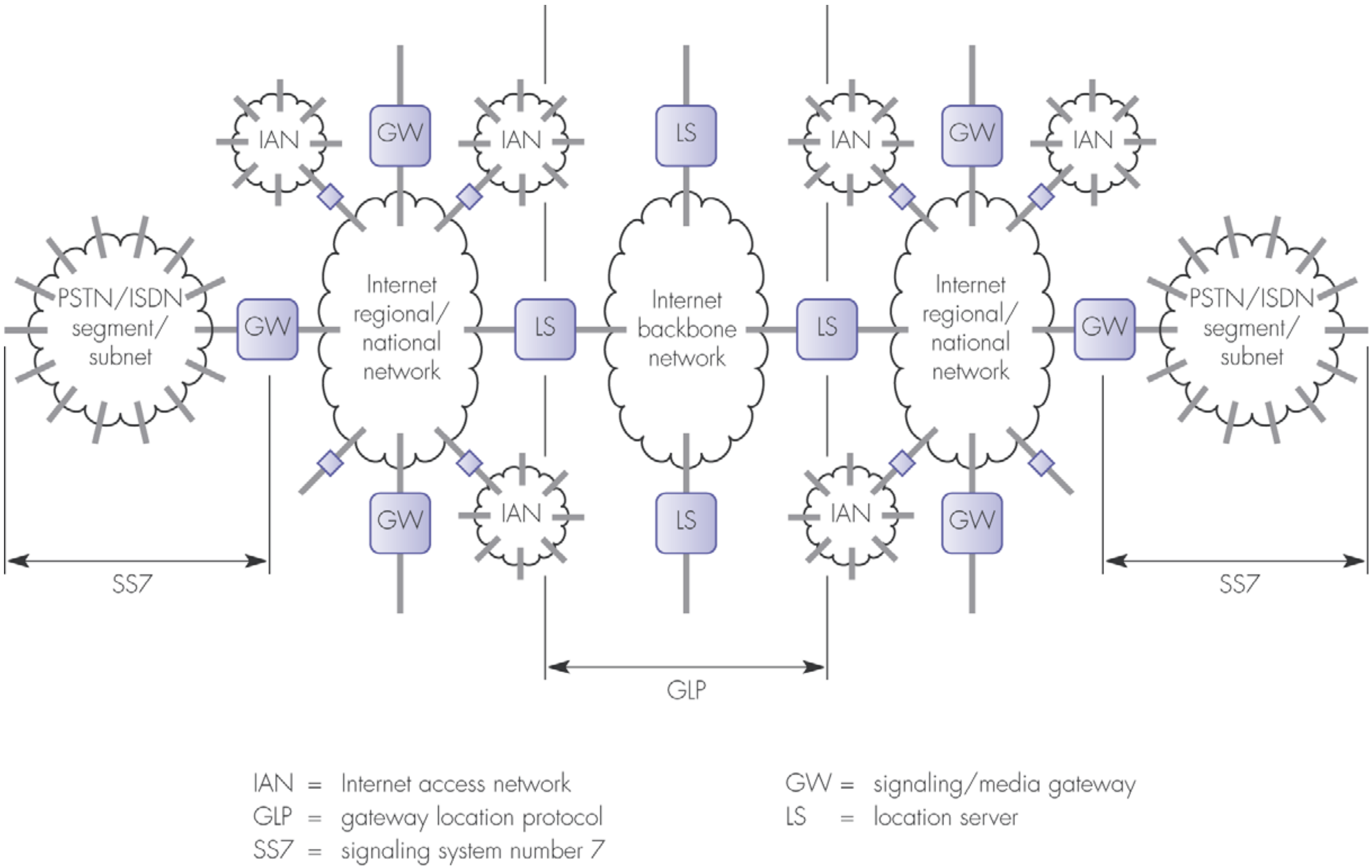

Figure 8.21 Interworking between Internet hosts and PSTN/ISDN terminal equipment

Skype

- A Peer-to-Peer (P2P) network for voice calls
- Uses that P2P network for ‘best’ routing of voice calls over UDP (or TCP if it can’t find a non-firewalled UDP path)
- Uses its own user namespace - skype usercodes
- Has a good audio codec of its own
- Has gateways to PSTN
- Cheap and widely used

but …

- Proprietary!