Punkt-till-punkt-access

- HDLC
  (High-level Data Link Control)
- PPP
  (Point-to-Point Protocol)

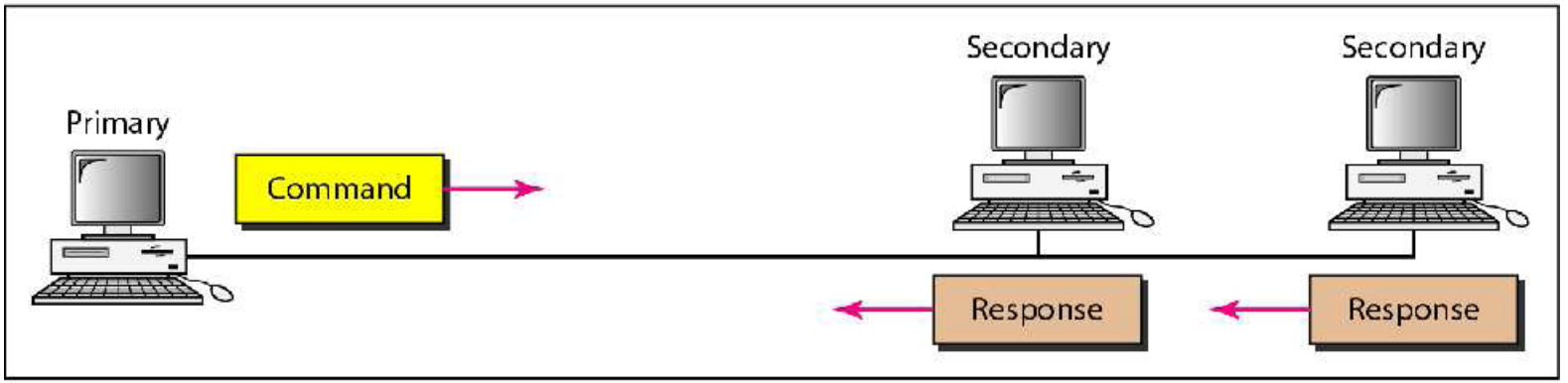
HDLC

- HDLC (High-level Data Link Control)
  - Används både till punkt-till-punkt och till multi-punkt-förbindelser
  - Använder ARQ (Automatic Repeat reQuest)

HDLC

- NRM (Normal Response Mode)

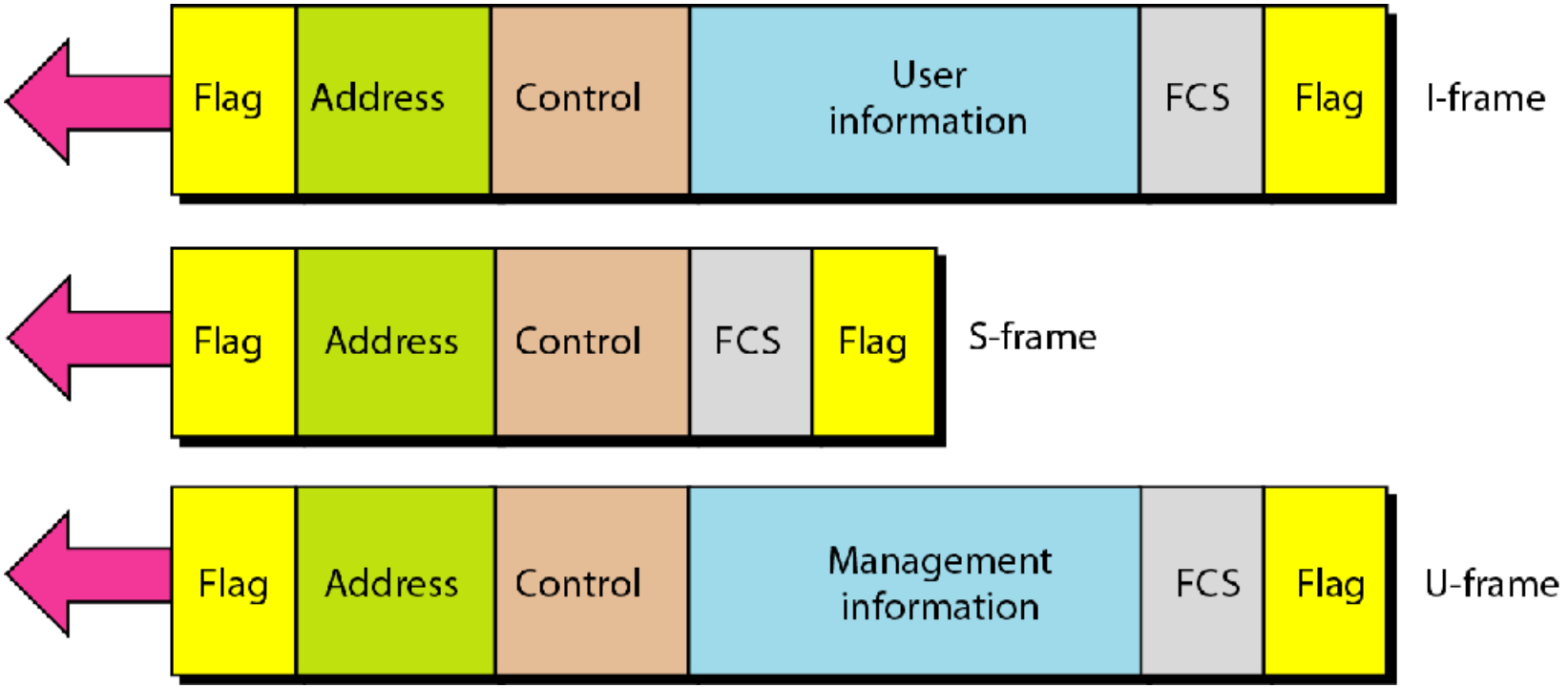
a. Point-to-point



b. Multipoint

HDLC

- De tre olika ramtyperna i HDLC



HDLC

- Flaggfält (Flag Field)
  - 8 bitar för synkronisering
- Adressfält (Address Field)
- Styrfält (Control Field)
- Informationsfält (Information Field)
  - Användar- eller systeminformation
- Ramkontrollfält (Frame Check Sequence Field)

HDLC

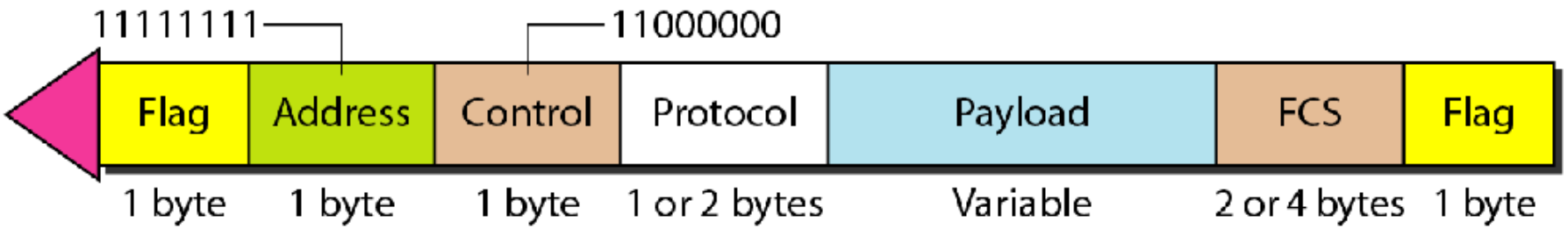
- I-frames (Information frames)
  - Förmedlar användardata från nätverkslagret
  - Kan innehålla information om flödes- och felkontroll (piggybacking)
- S-frames (Supervisor frames)
  - För flödes- och felkontrollinformation
- U-frames (Unnumbered frames)
  - Förmedlar system- och styrinformation

PPP

- PPP (Point-to-Point Protocol)
  - Vanligaste protokollet för punkt-till-punkt-förbindelser
  - Används för kontakt mellan användare och internetleverantör
  - Använder en variant av HDLC

PPP

- Formatet på en PPP-ram

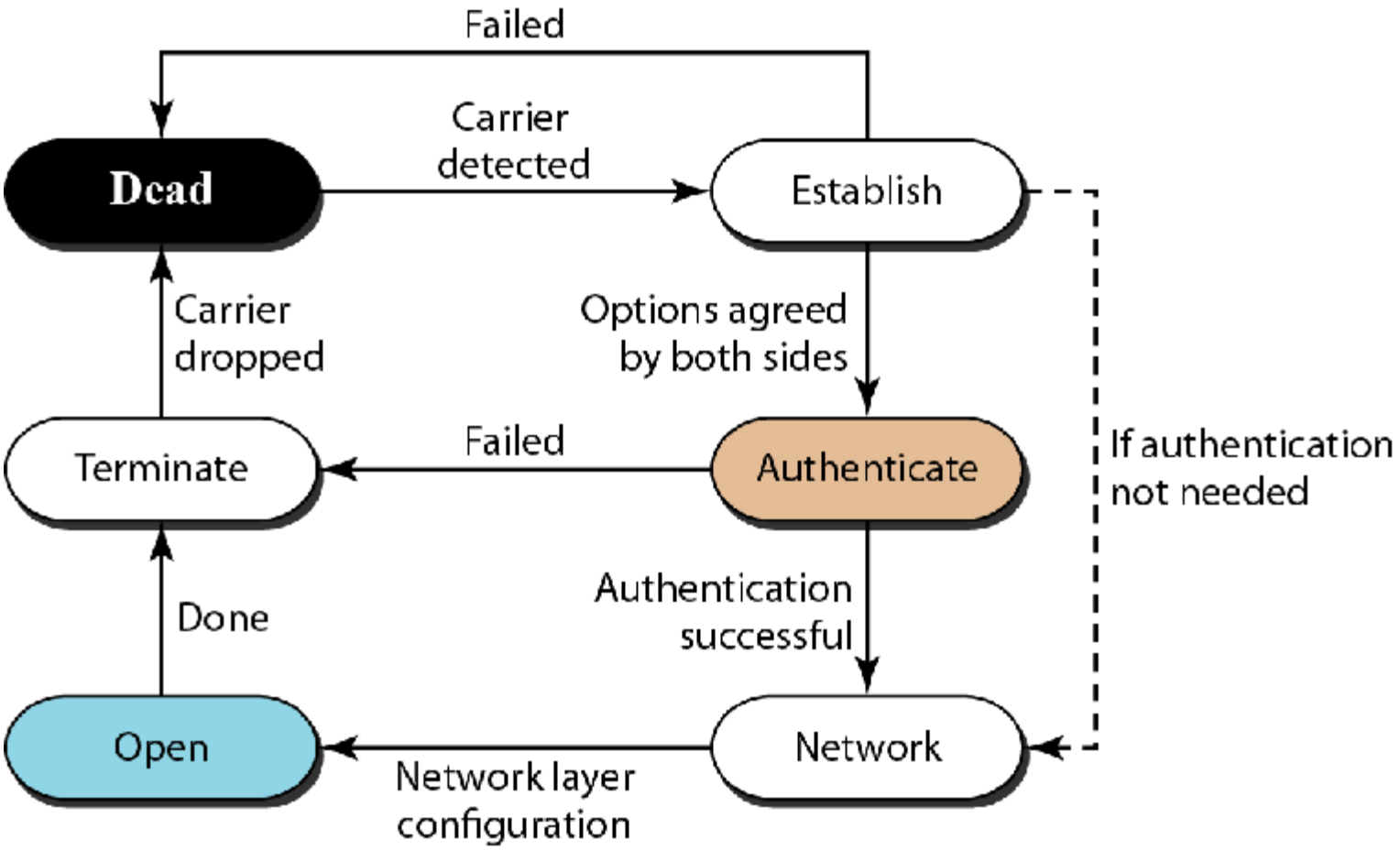


PPP

- Fälten i en PPP-ram
  - Flaggfälten för avgränsning (alltid 01111110 i PPP)
  - Adressfältet (HDLC:s broadcast 11111111)
  - Styrfältet (alltid 11000000)
  - Protokollfältet
  - Datafältet
  - FCS-fältet (CRC)

PPP

- Tillståndsgraf för PPP

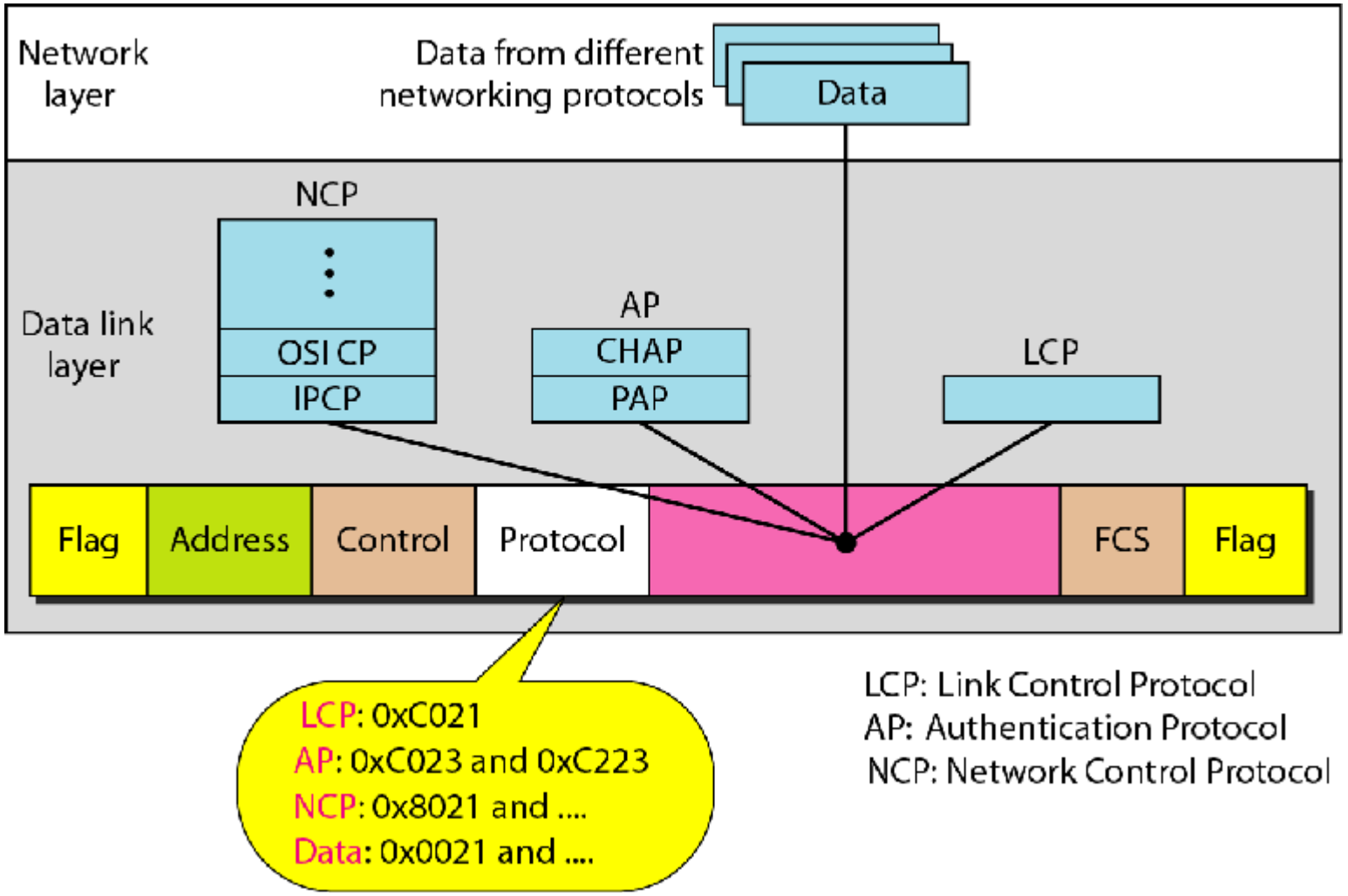


PPP

- Protokollstack för PPP
  - LCP (Link Control)
  - PAP (Password Authentification)
  - CHAP (Challenge Handshake Authentification)

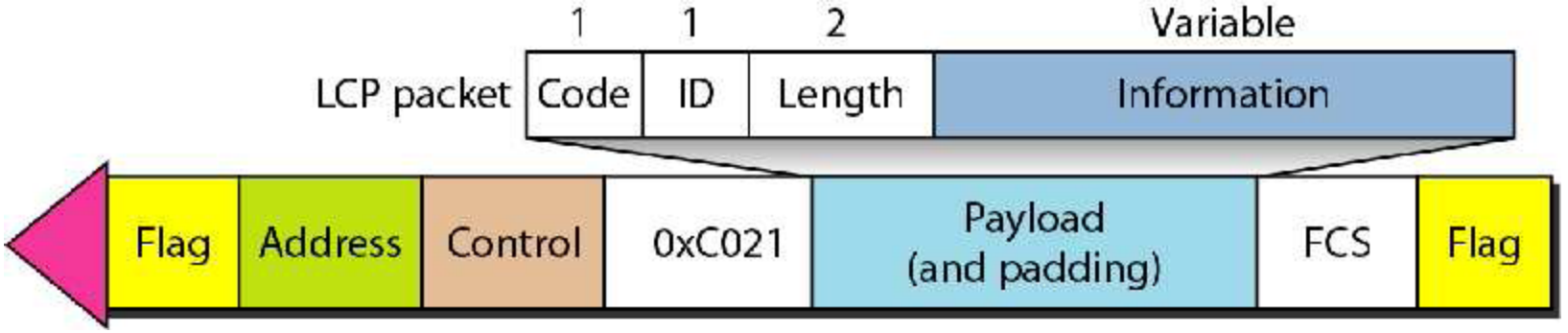
PPP

- Flera lager av protokoll



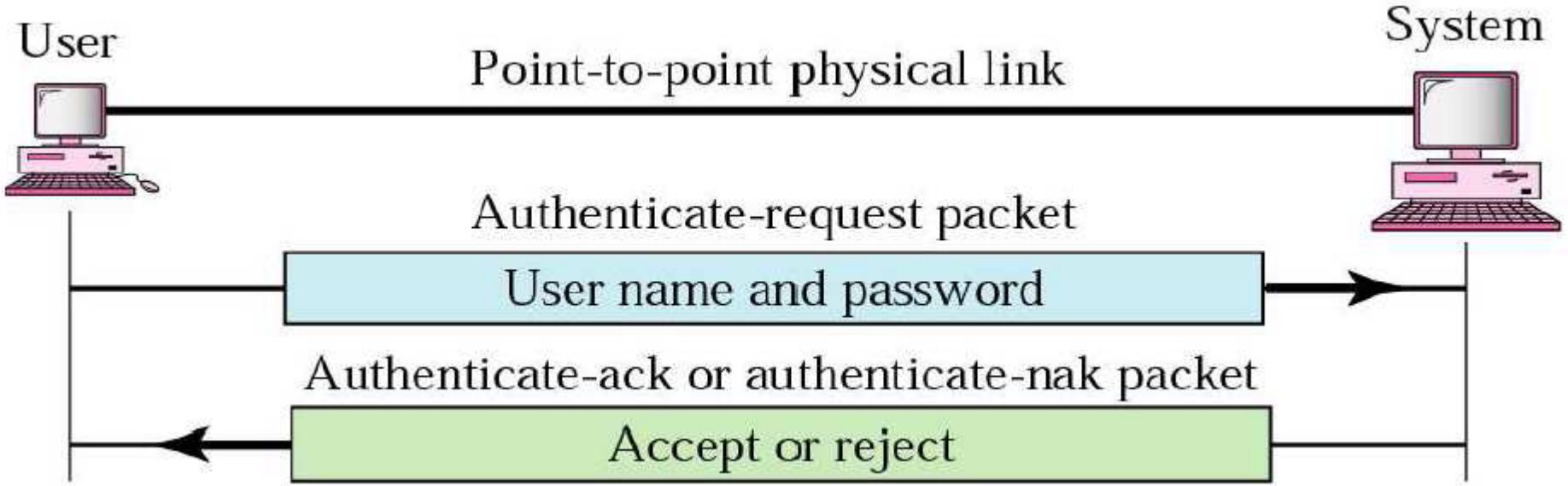
PPP

- LCP-paket i en PPP-ram
  - Kod (Code): typen av LCP-paket
  - ID: för matchning av förfrågan och svar
  - Längd (Length): Totala LCP-paketets längd
  - Information: extra informaton



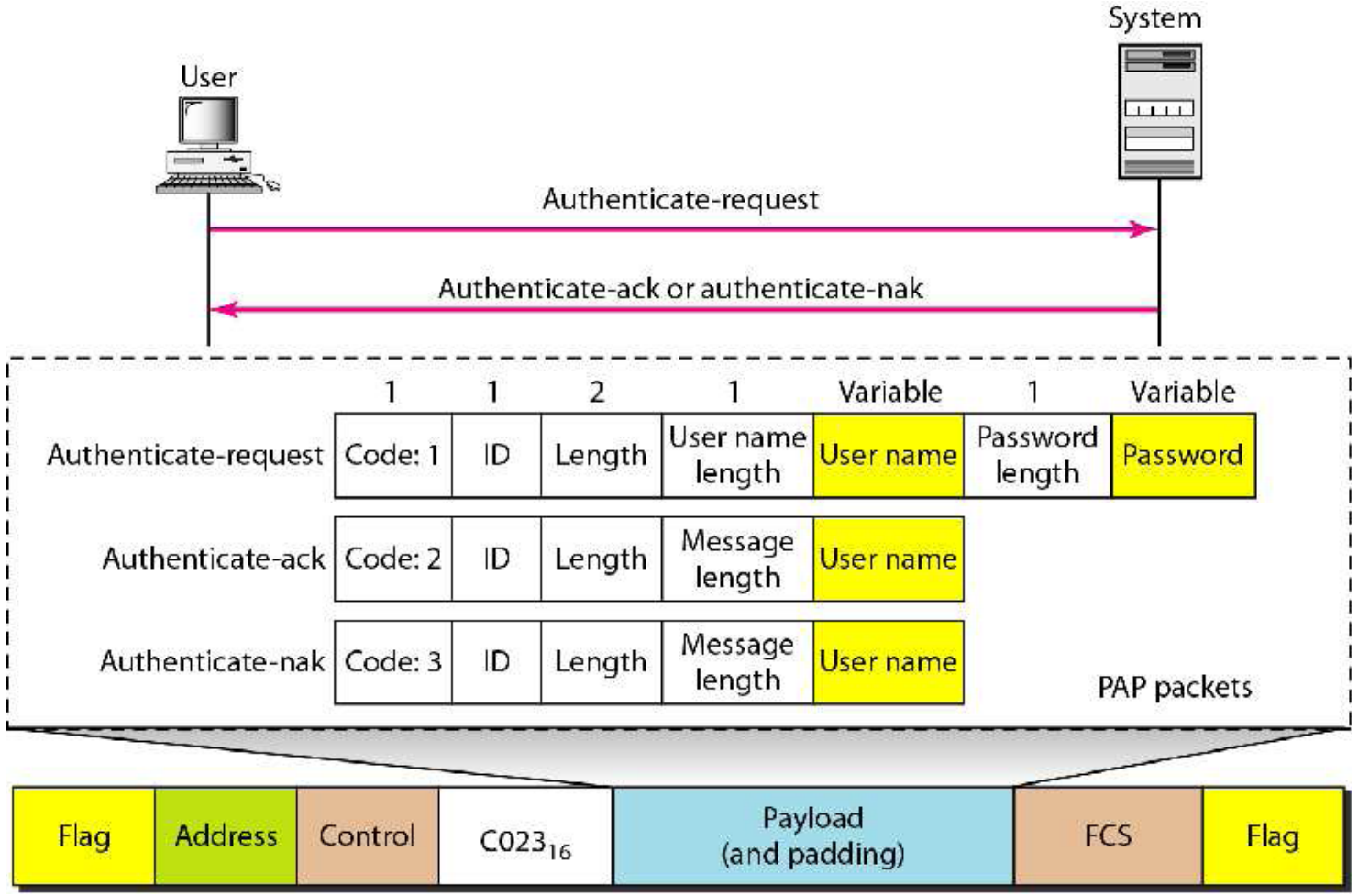
PPP

- PPP-kommunikation med PAP-autentisering



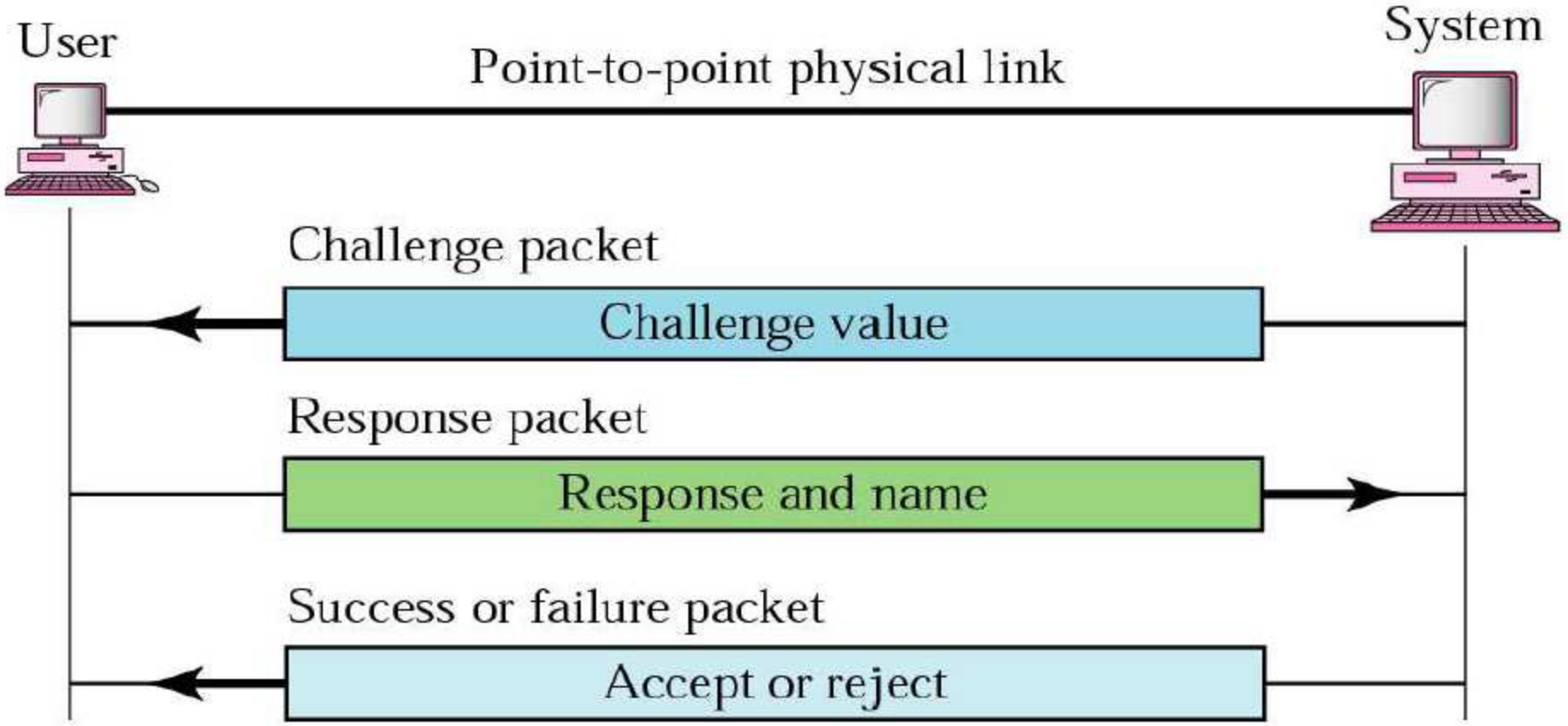
PPP

- PAP-paket i PPP-ram



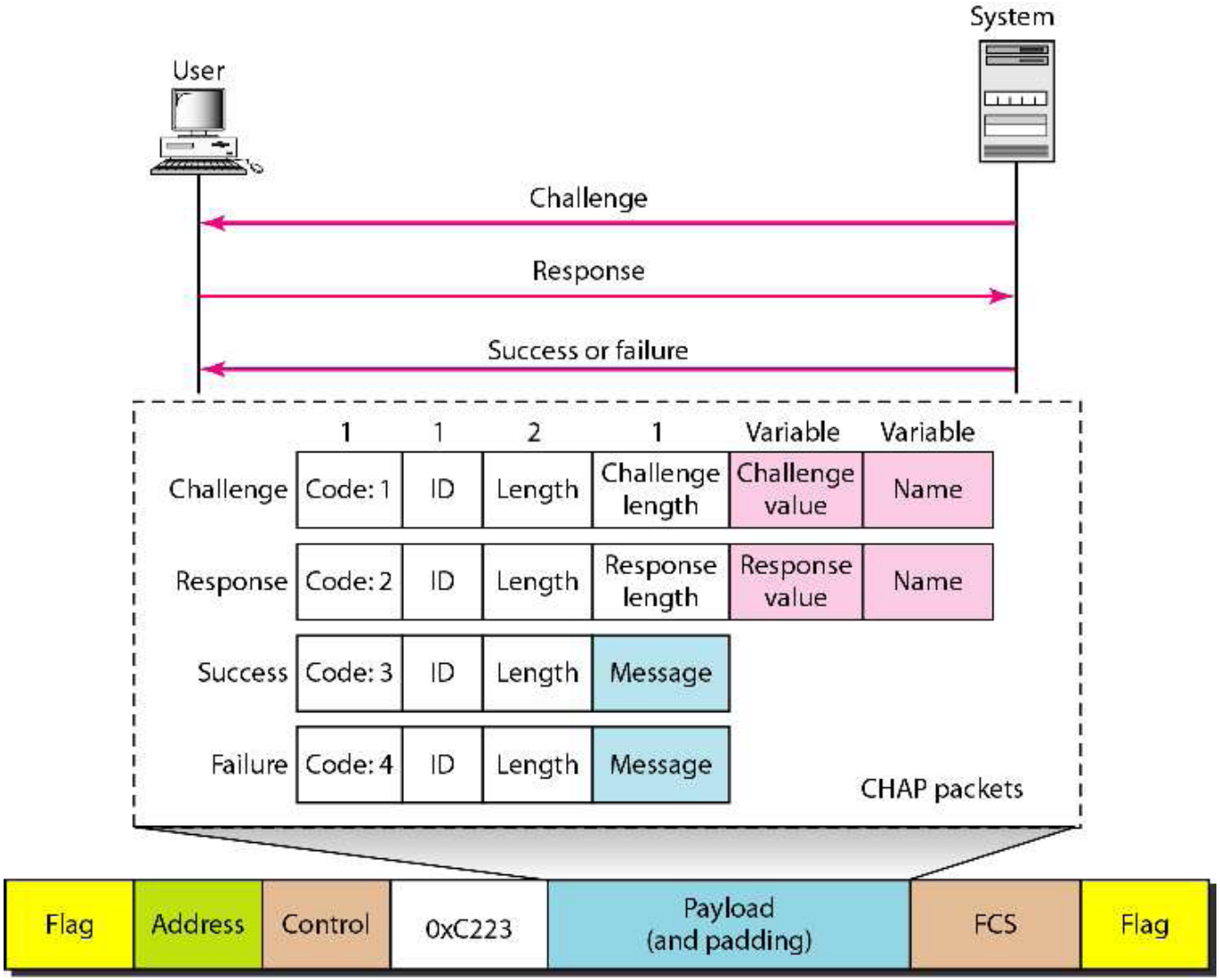
PPP

- PPP-kommunikation med CHAP-autentisering



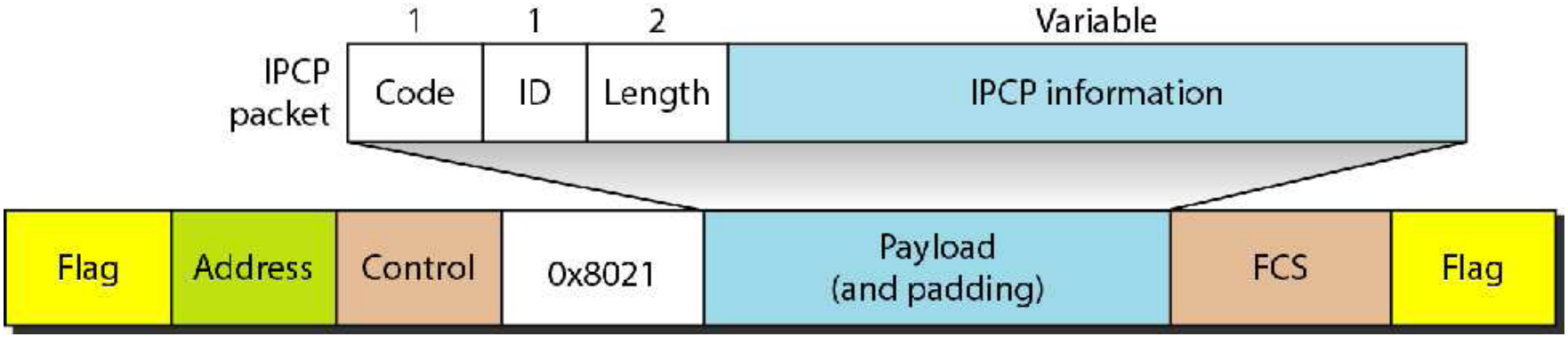
PPP

- CHAP-paket i PPP-ram



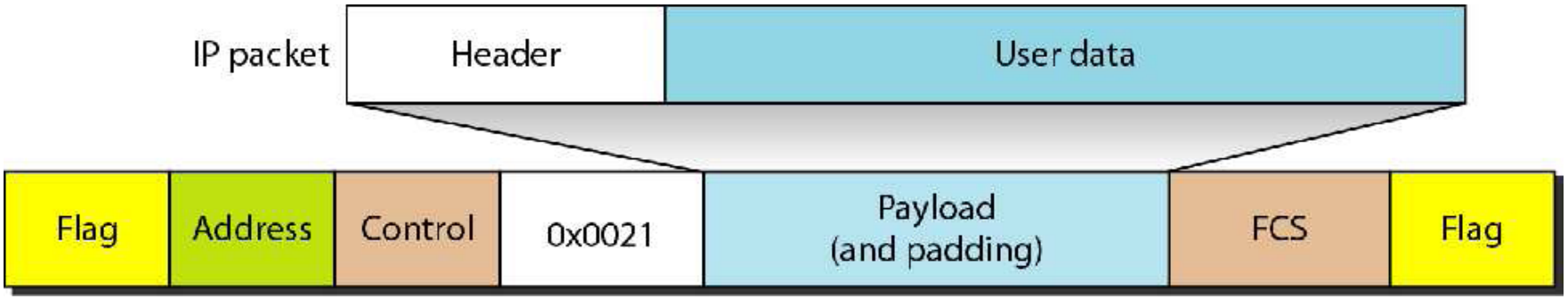
PPP

- IPCP-paket i PPP-ram

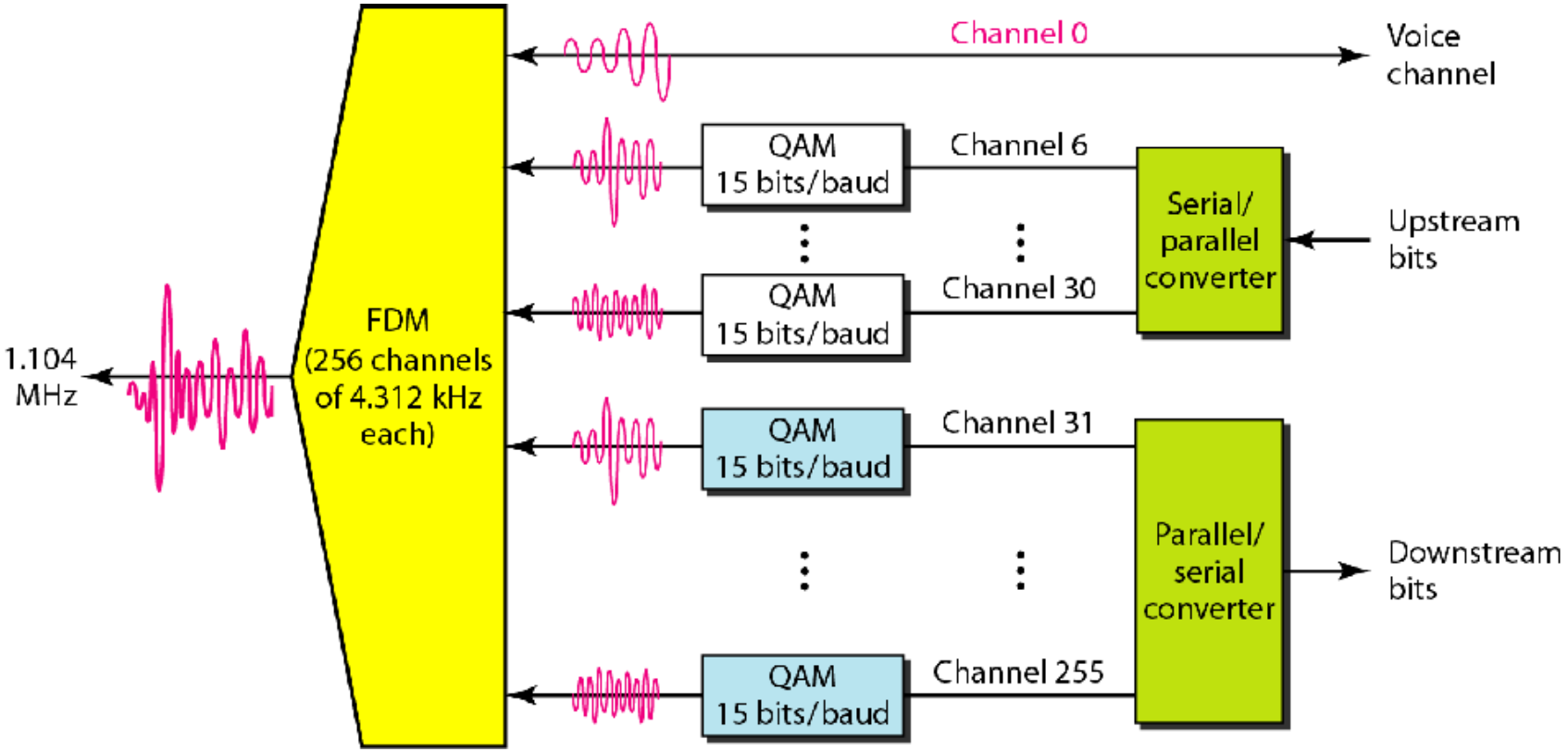


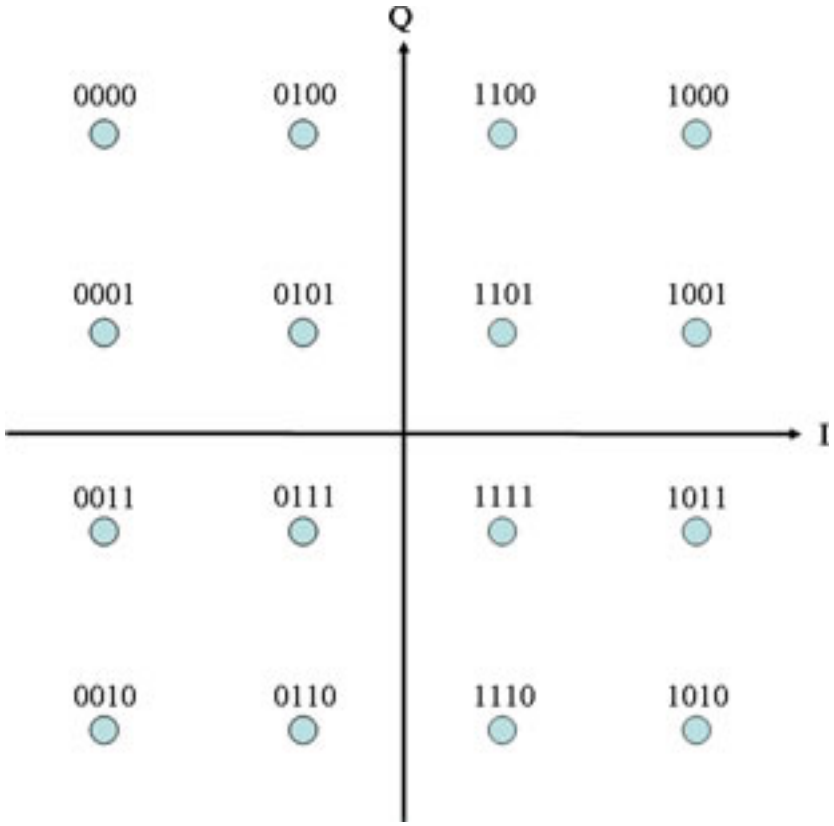
PPP

- IP-paket i PPP-ram



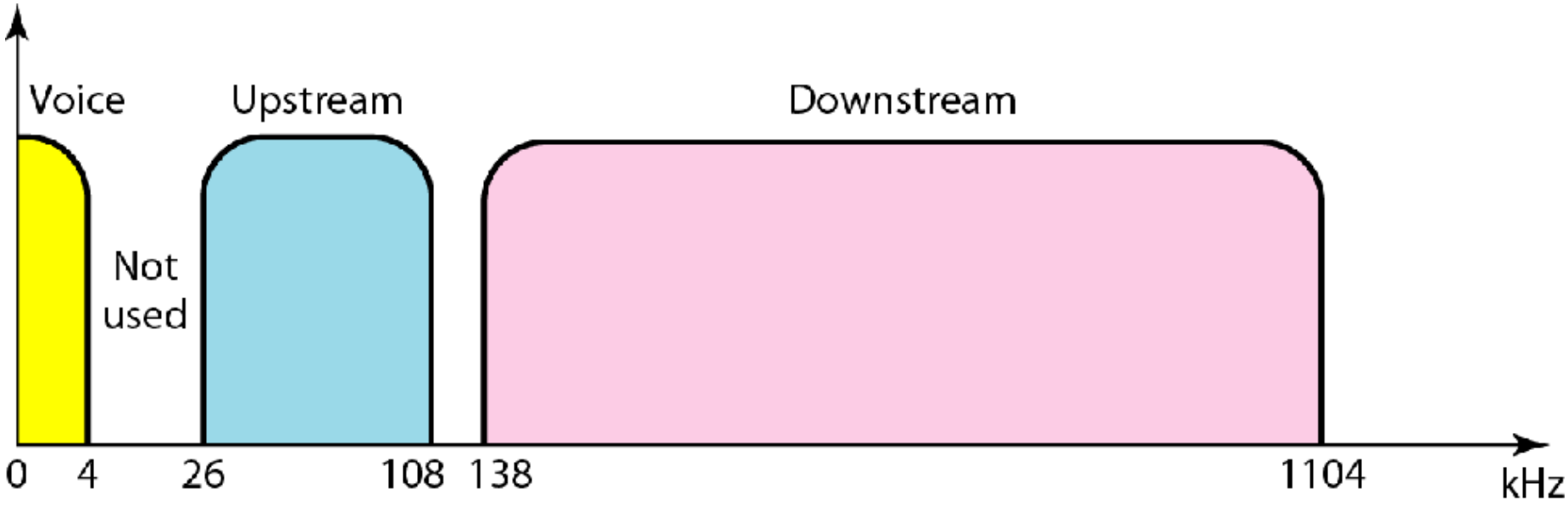
ADSL


1.104
MHz
FDM
(256 channels
of 4.312 kHz
each)
Channel 0
Voice
channel
QAM
15 bits/baud
Channel 6
Serial/
parallel
converter
Upstream
bits
QAM
15 bits/baud
Channel 30
QAM
15 bits/baud
Channel 31
Parallel/
serial
converter
Downstream
bits
QAM
15 bits/baud
Channel 255


Q
I
0000
0100
1100
1000
0001
0101
1101
1001
0011
0111
1111
1011
0010
0110
1110
1010

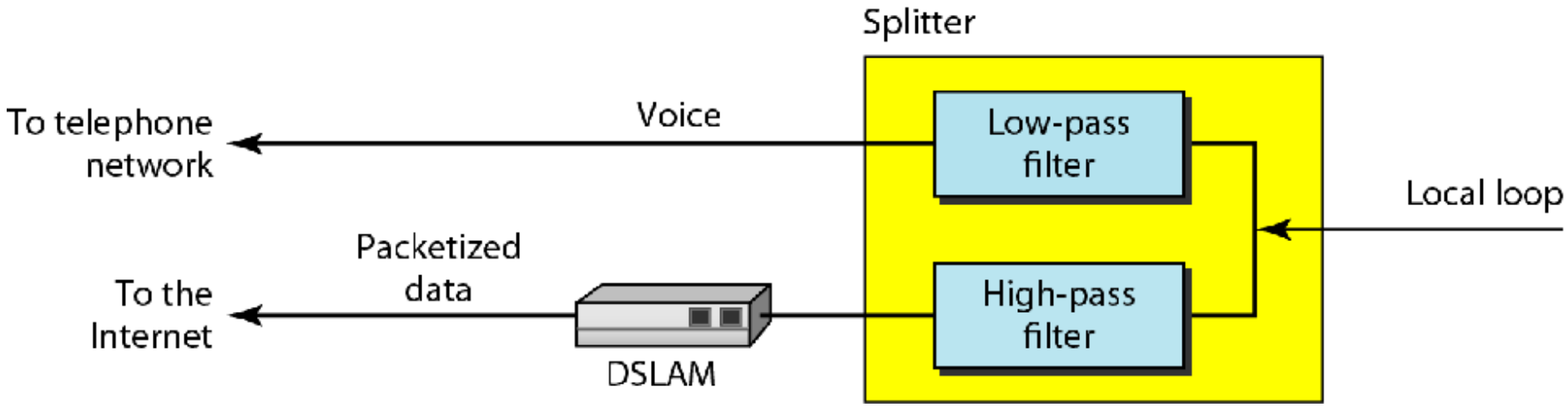
ADSL

- Bandbreddsutnyttjande vid ADSL
  - Assymmetri mellan uppströms och nedströms



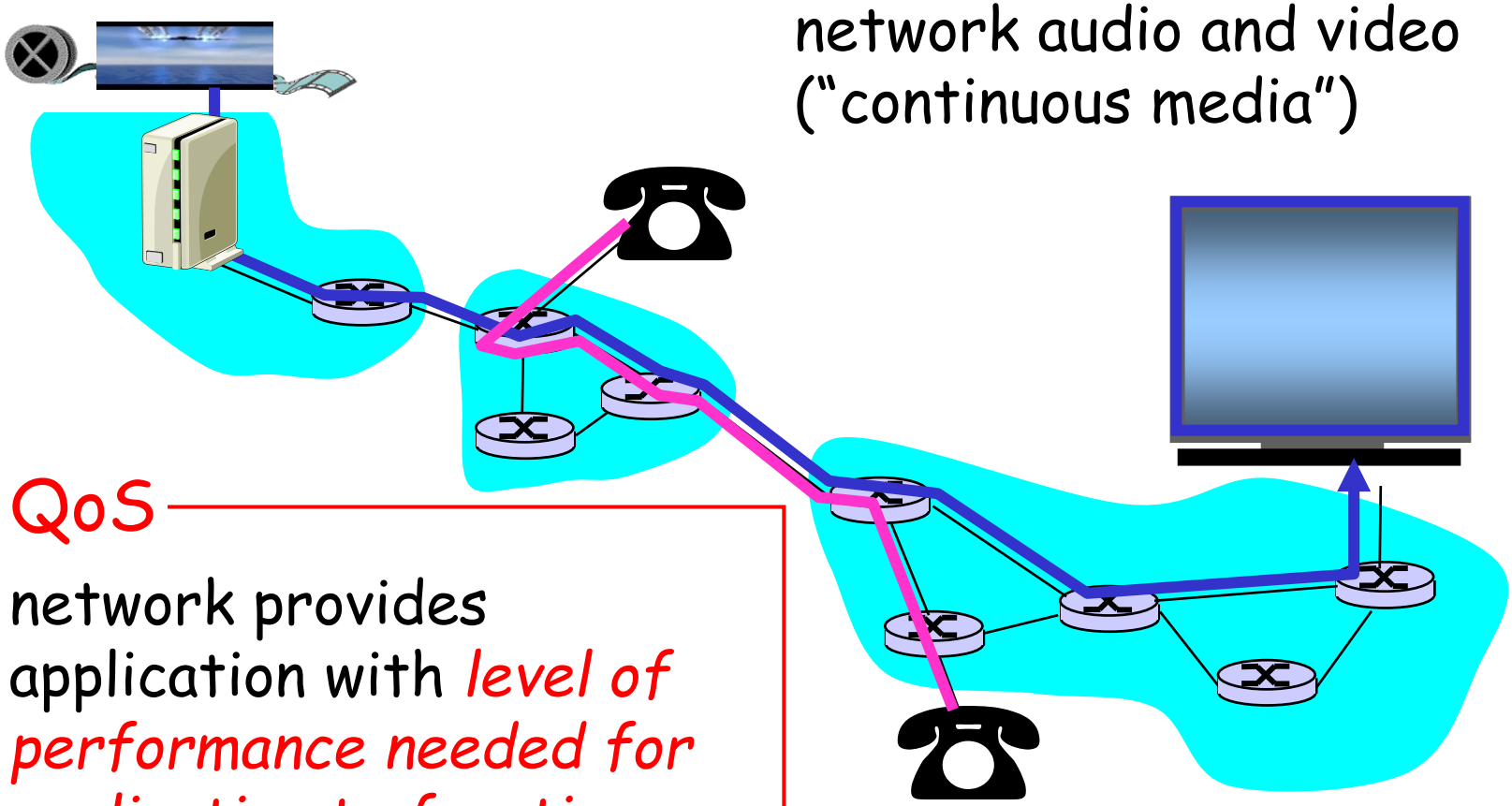
ADSL

- Användning av s.k. splitter för uppdelning mellan tal och data genom filtrering



Multimedia and Quality of Service: What is it?

multimedia applications:
network audio and video
("continuous media")



QoS

network provides application with *level of performance needed for application to function.*

MM Networking Applications

Classes of MM applications:

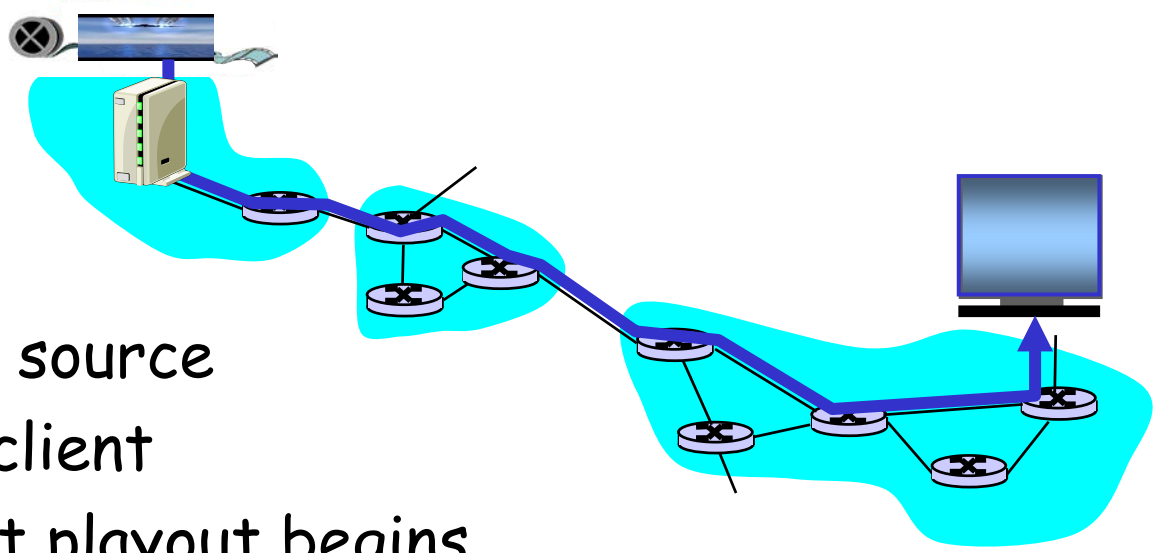
1) stored streaming
2) live streaming
3) interactive, real-time

Jitter is the variability of packet delays within the same packet stream

Fundamental characteristics:

- typically **delay sensitive**
  - end-to-end delay
  - delay jitter
- **loss tolerant**: infrequent losses cause minor glitches
- antithesis of data, which are loss *intolerant* but delay *tolerant*.

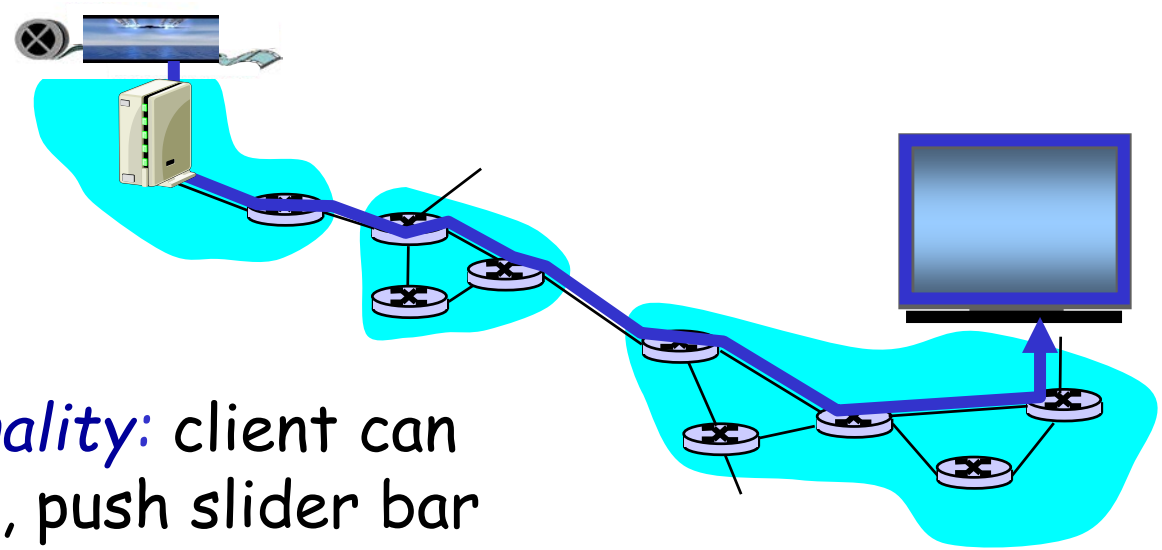
Streaming Stored Multimedia



Stored streaming:

- media stored at source
- transmitted to client
- *streaming:* client playout begins *before* all data has arrived
  - timing constraint for still-to-be transmitted data: in time for playout

Streaming *Stored* Multimedia: Interactivity



- *VCR-like functionality:* client can pause, rewind, FF, push slider bar
  - 10 sec initial delay OK
  - 1-2 sec until command effect OK

- timing constraint for still-to-be transmitted data: in time for playout

Streaming *Live* Multimedia

Examples:

- Internet radio talk show
- live sporting event

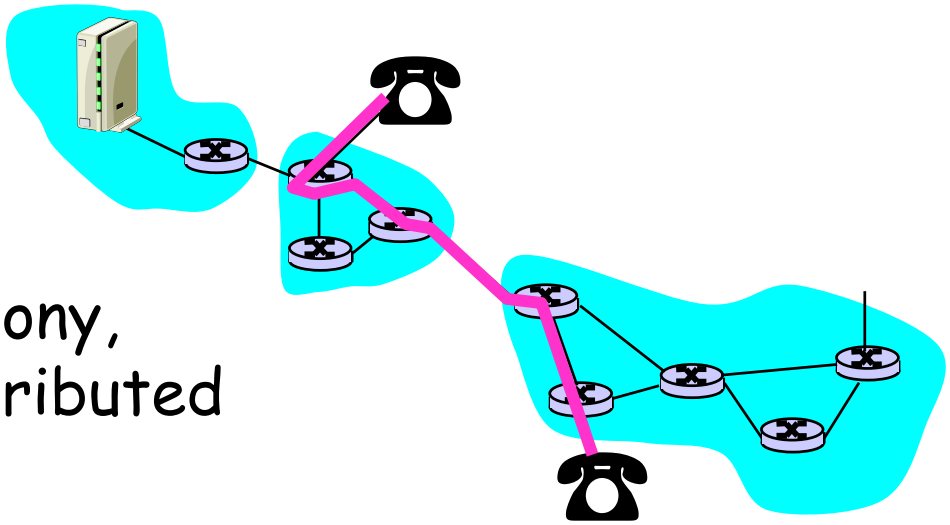
Streaming (as with streaming *stored* multimedia)

- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity

- fast forward impossible
- rewind, pause possible!

Real-Time Interactive Multimedia



- applications: IP telephony, video conference, distributed interactive worlds
- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetization) and network delays
    - higher delays noticeable, impair interactivity
- session initialization
  - how does caller advertise its IP address, port number, encoding algorithms?

Multimedia Over Today's Internet

TCP/UDP/IP: "best-effort service"

- *no* guarantees on delay, loss

But you said multimedia apps requires QoS and level of performance to be effective!

Today's Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss

A few words about video compression

- video: sequence of images displayed at constant rate
  - e.g. 24 images/sec
- digital image: array of pixels
  - each pixel represented by bits
- redundancy
  - spatial (within image)
  - temporal (from one image to next)

Examples:

- MPEG 1 (CD-ROM) 1.5 Mbps
- MPEG2 (DVD) 3-6 Mbps
- MPEG4 (often used in Internet, < 1 Mbps)

Research:

- layered (scalable) video
  - adapt layers to available bandwidth

Streaming Stored Multimedia

application-level streaming techniques for making the best out of best effort service:

- client-side buffering
- use of UDP versus TCP
- multiple encodings of multimedia

Media Player

- jitter removal
- decompression
- error concealment
- graphical user interface w/ controls for interactivity

Streaming Multimedia: UDP or TCP?

UDP

server sends at rate appropriate for client (oblivious to network congestion !)

- often send rate = encoding rate = constant rate
- then, fill rate = constant rate - packet loss

short playout delay (2-5 seconds) to remove network jitter

error recover: time permitting

TCP

send at maximum possible rate under TCP

fill rate fluctuates due to TCP congestion control

larger playout delay: smooth TCP delivery rate

HTTP/TCP passes more easily through firewalls

Summary: Internet Multimedia: bag of tricks

use UDP to avoid TCP congestion control (delays) for time-sensitive traffic

client-side adaptive playout delay: to compensate for delay

server side matches stream bandwidth to available client-to-server path bandwidth

- chose among pre-encoded stream rates
- dynamic server encoding rate

error recovery (on top of UDP)

- FEC, interleaving, error concealment
- retransmissions, time permitting

CDN (Content Distribution Network):

bring content closer to clients