



# Voice/IP




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# ... Voice and Data

- Analog voice transmission has dominated the communications industry for the past 100 years
  - The entrance of multi-service digital networks is placing a new set of demands on the service profile of communications networks
  - Will we see convergence to a single network platform?
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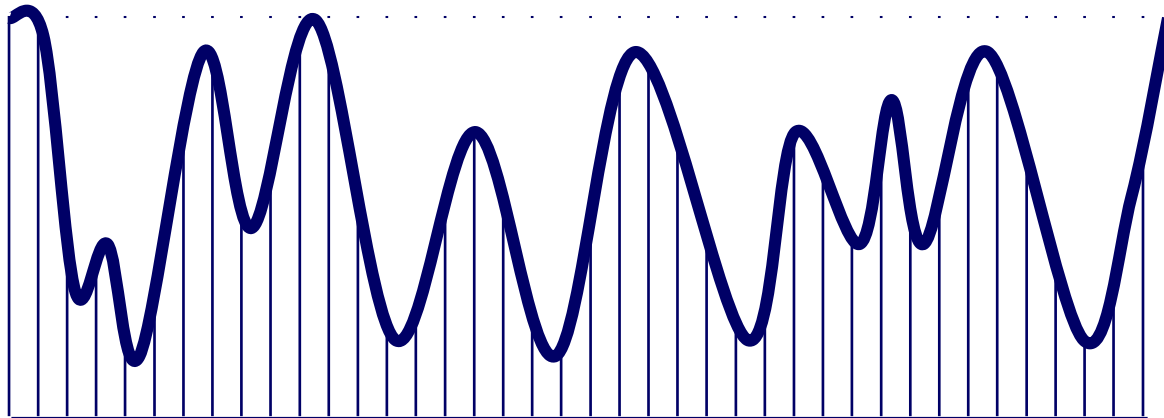


# Voice Networks

- Voice transmissions have an number of characteristics:
    - 3Khz bandwidth
    - limited amplitude (<25db)
    - time synchronization
    - limited average duration (200 seconds)
    - High localization (80:20 rule)
    - Strong traffic peaking characteristics
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# Digitizing Voice

- 8000 samples per second (Nyquist Theorem)
  - 125  $\mu$ second timebase
- 256 discrete amplitude levels
  - 8 bits per sample
- 64Kbps PCM data stream






# 64K Networks

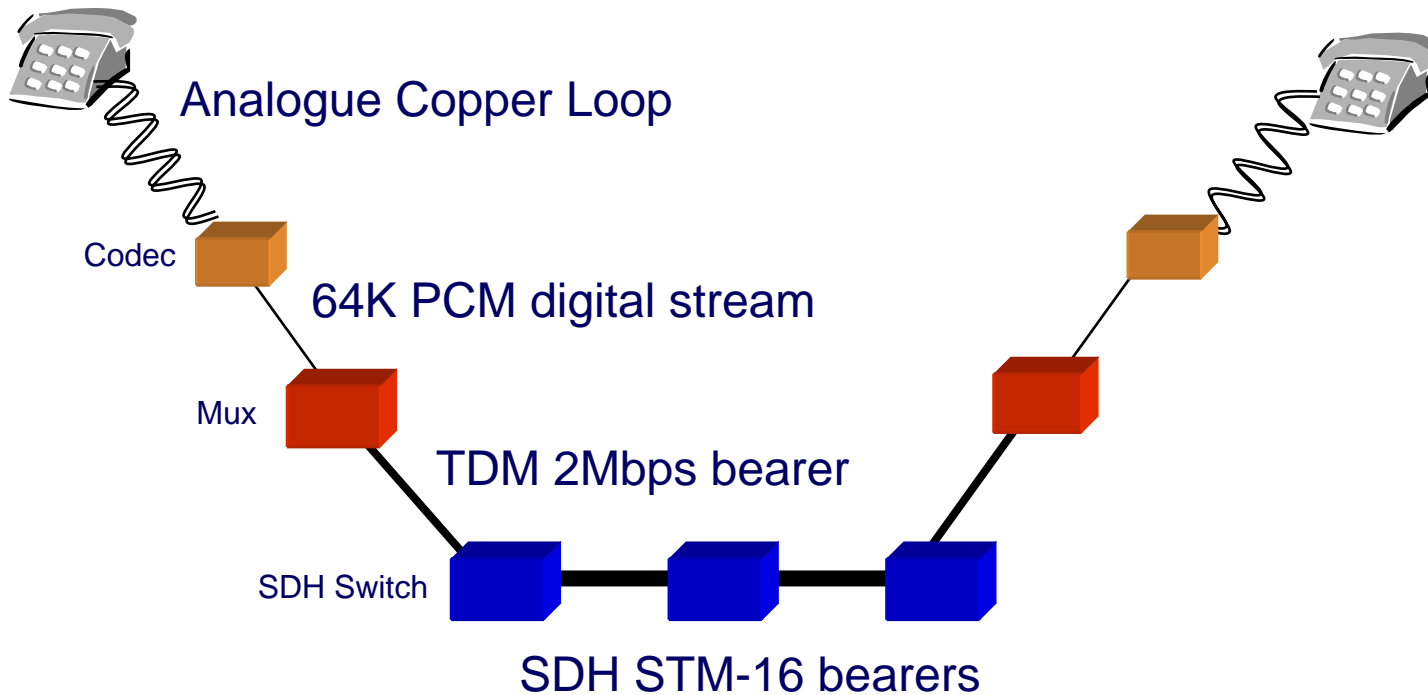
- Voice networks are built by multiplexing and switching synchronous 64K data streams
  - Time division multiplexing
    - 125  $\mu$ second time base
    - 8 bit symbols per time slot per voice channel
  - 2Mbps bearer is 32 x 64K slots
    - 30 data slots
    - 1 channel signaling slot
    - 1 frame sync slot
    - = 2048Mbps
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# Circuit-switched Networks


- Time division switches
    - reorder the timeslots of a TDM data stream
    - impose 1 slot time constant delay
  - Space Switches
    - crossbar switching
    - 2 slot time delay due to muxing overhead
  - Supports dynamically switchable end-to-end synchronously clocked circuits
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# A Voice Network






# A Data Network

- Switches *Packets*, not *circuits*
  - Each packet may be independently forwarded, delayed or dropped by each router
  - Each packet is independently switched to its addressed destination
  - There is no time synchronization between sender and receiver
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# Data Networks

- Highly cost effective infrastructure
    - low levels of network functionality
    - high potential carriage efficiency
  - Functionality pushed beyond the network edge
  - Assumption of adaptive data flow control by end hosts
  - No guarantees of service level by the network.
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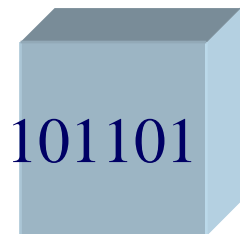
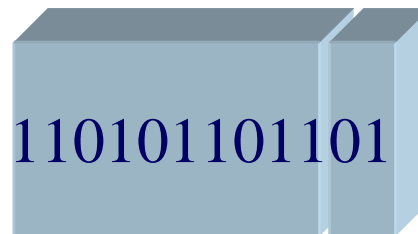
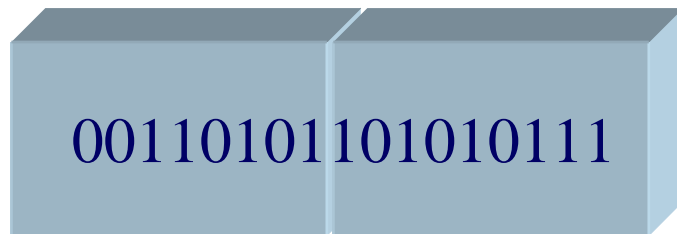
# ... Voice over IP

- packetize the digital voice stream
  - add timing information
  - add IP headers
  - send across the network
  - strip IP headers
  - feed into playback buffer using timing information
  - playback analogue signal
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# Packetizing Voice

- Compress the digital stream
  - differential PCM
  - Linear Predictive Encoding
  - silence suppression
- packetize the stream into fixed length payloads

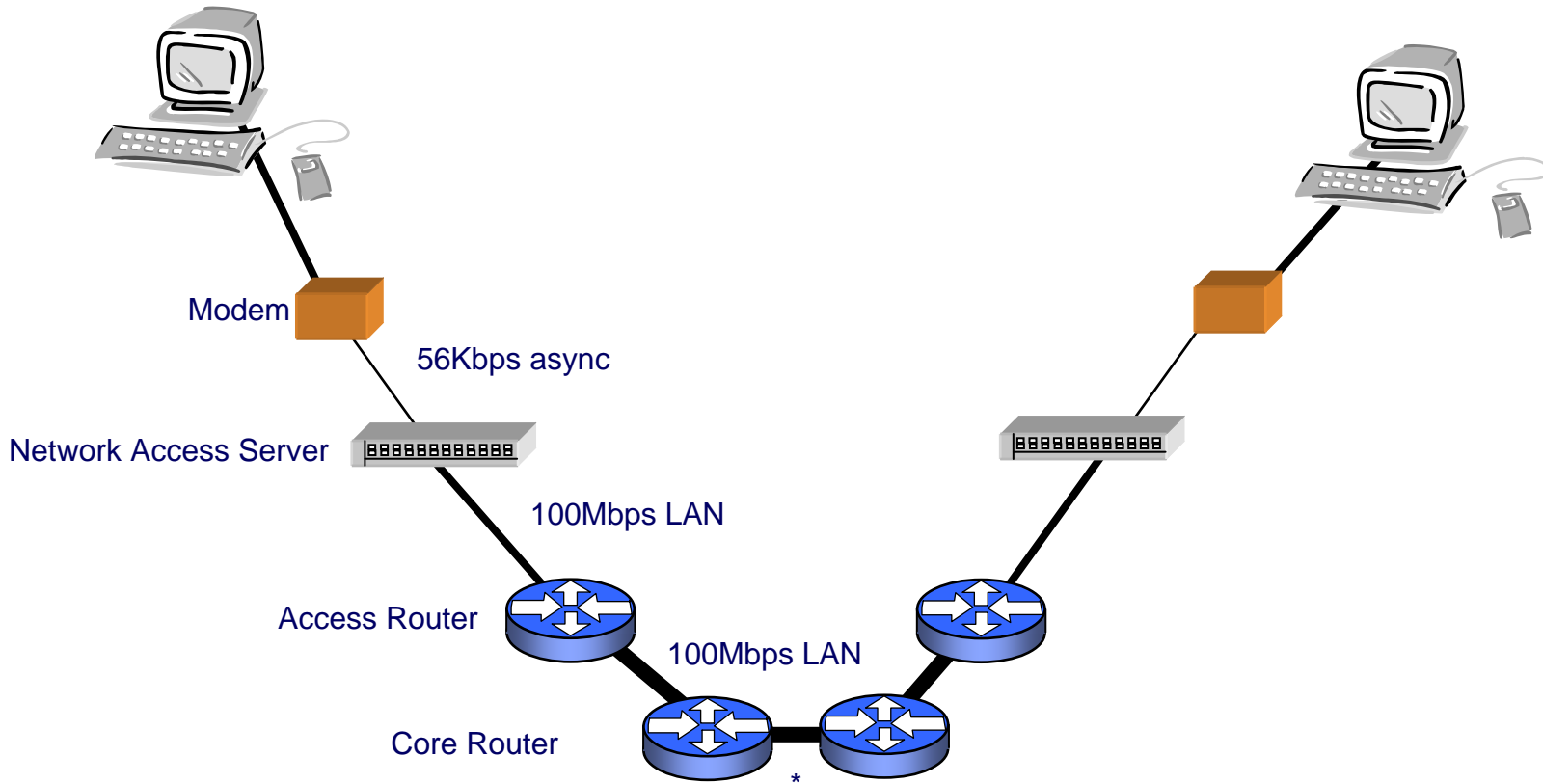


# ... Voice over IP

- Insert RTP header
  - 12 bytes or more
- Insert UDP header
  - 8 bytes
- Insert IP header
  - 20 bytes or more
- Payload size (packet rate) is a compromise between packet overhead and latency and jitter



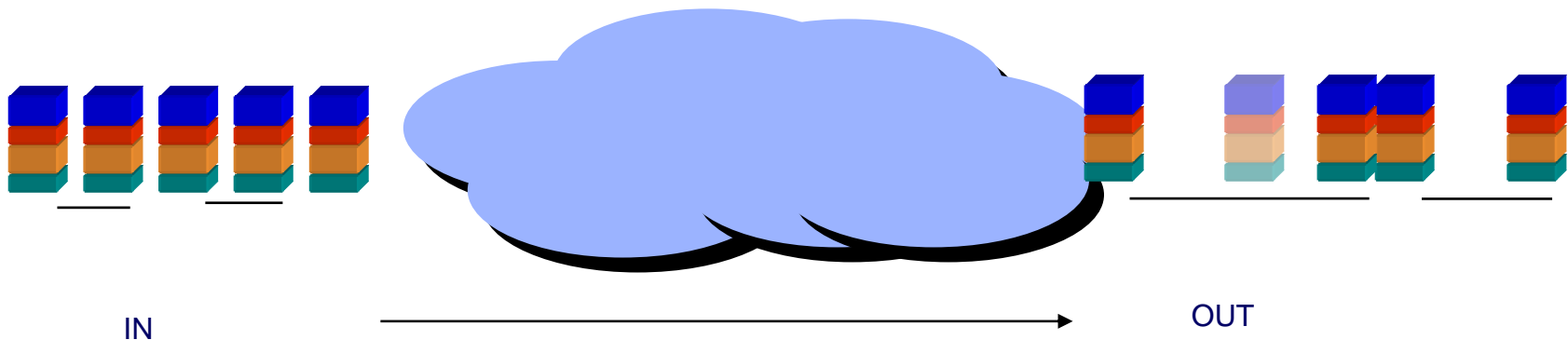
# Voice over IP



# ... VoIP Service Requirements

- Bounded End-to-End

- Delay - interaction requires delay to be under 500ms
- Jitter - high jitter causes large playback buffers
- Drop - signal quality

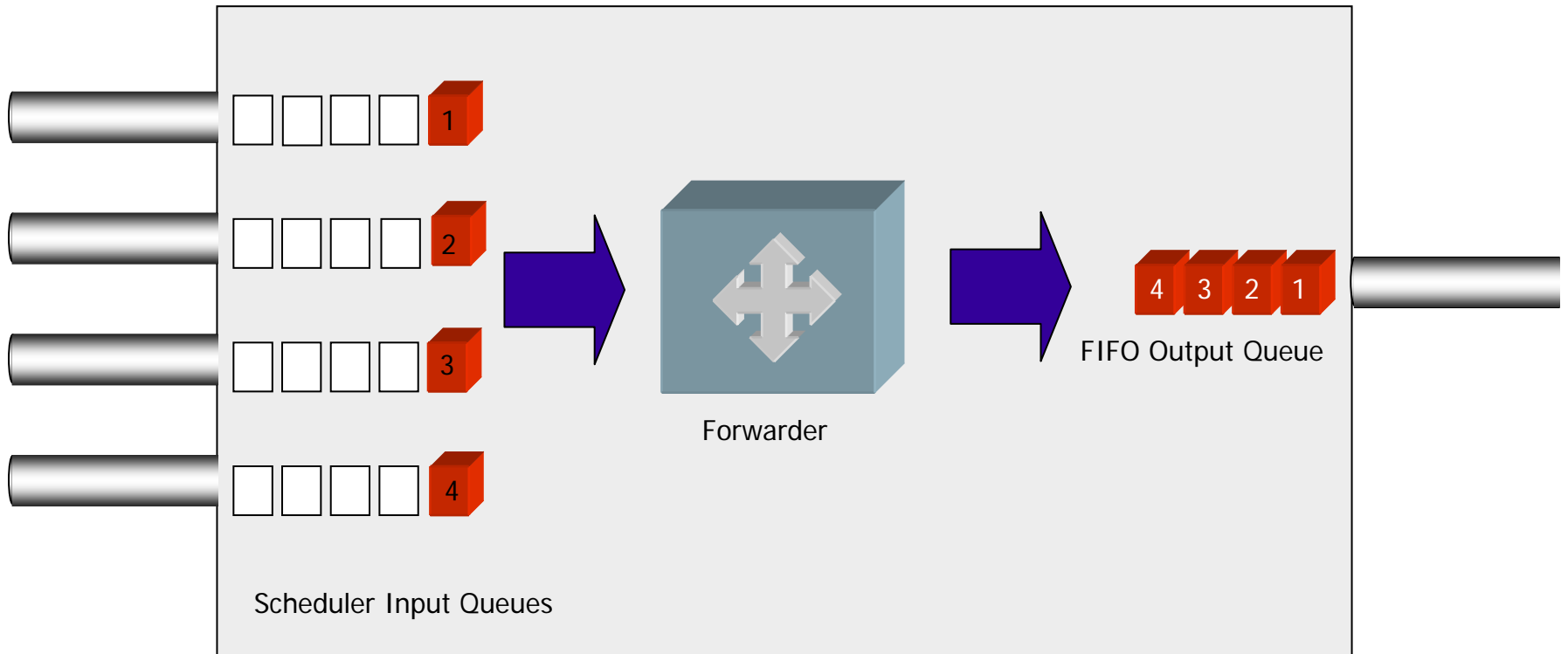




# Why do Routers have queues?

- Delay, Jitter and Drop are all outcomes of router queue behaviour
- Queues are used to:
  - resolve contention for a resource
  - buffer speed differences within the network


# Resource Contention Queues



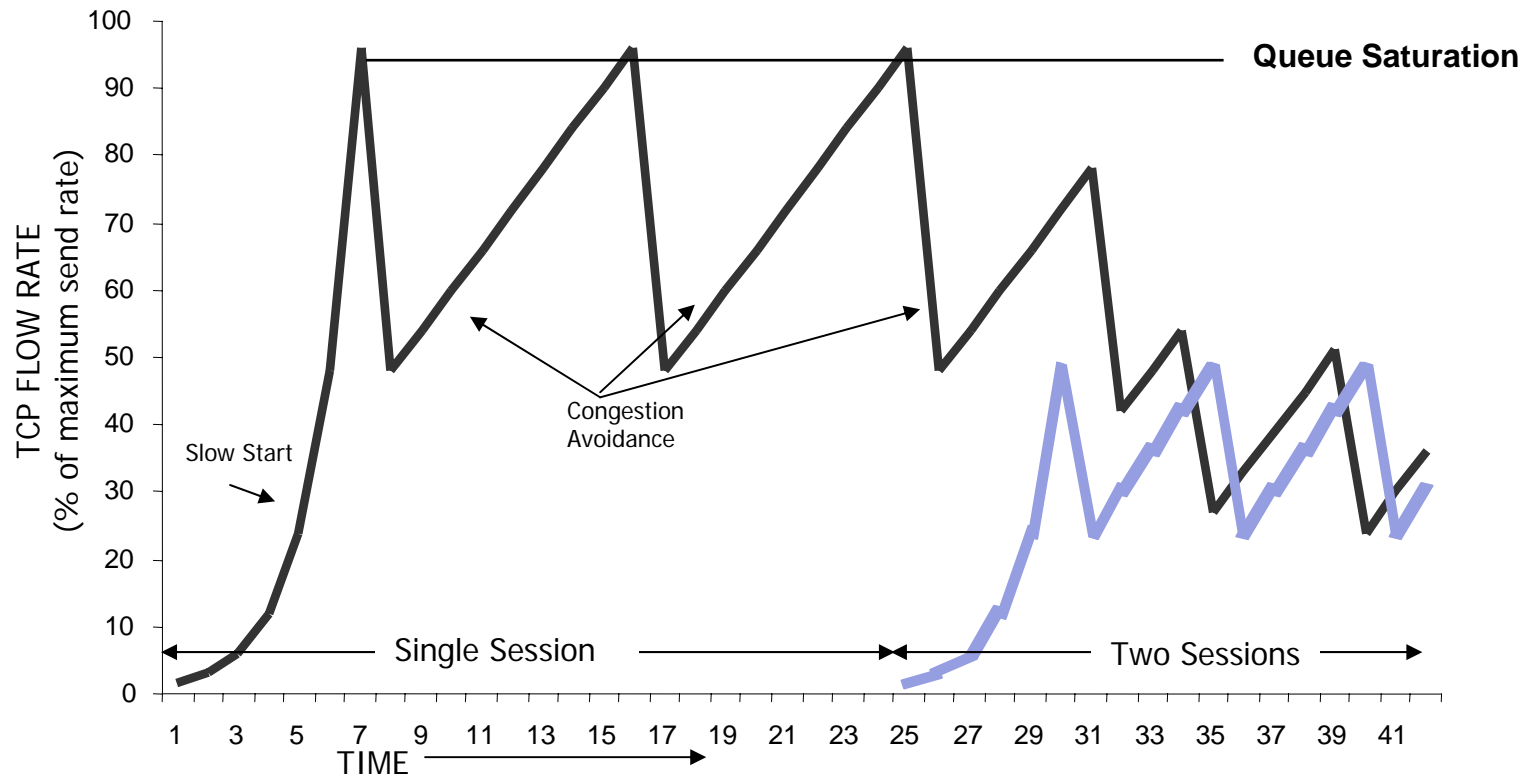




# ... TCP and Queues


- TCP is an adaptive data protocol
  - TCP has no 'fixed' data transfer rate.
  - Instead, TCP uses an adaptive flow control algorithm
  - TCP uses a feedback loop to adjust the sending rate to the available network capacity
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# TCP rate control

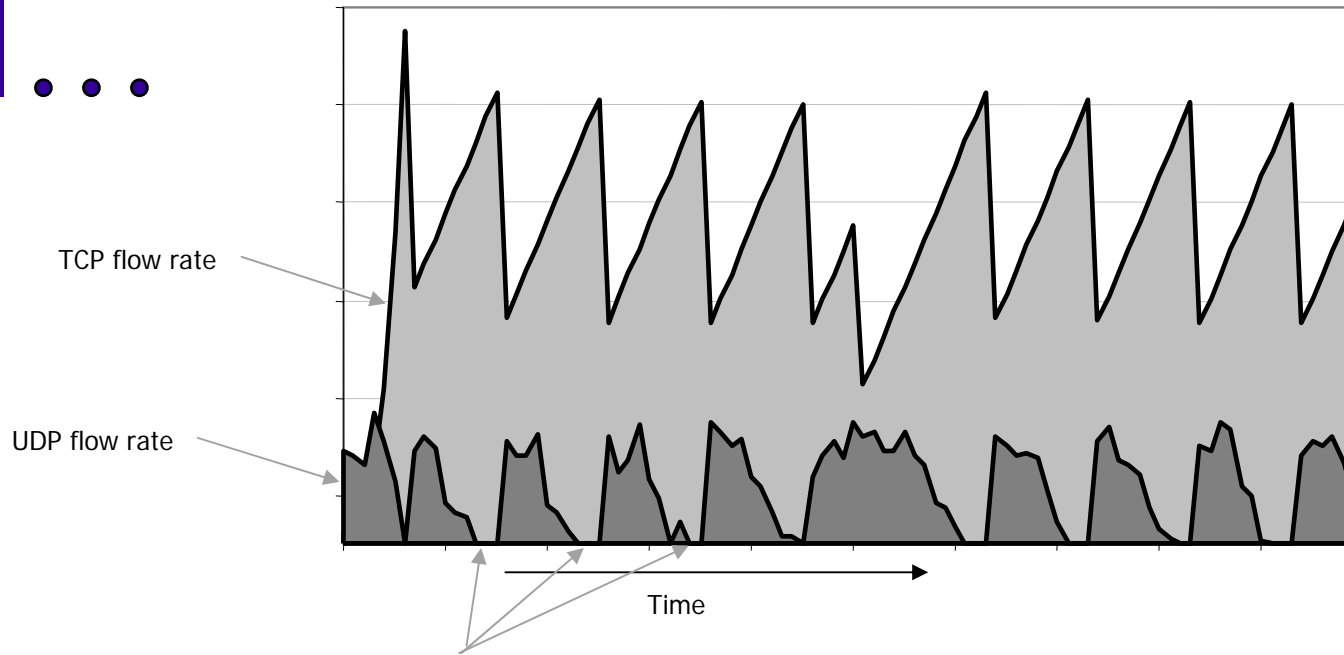




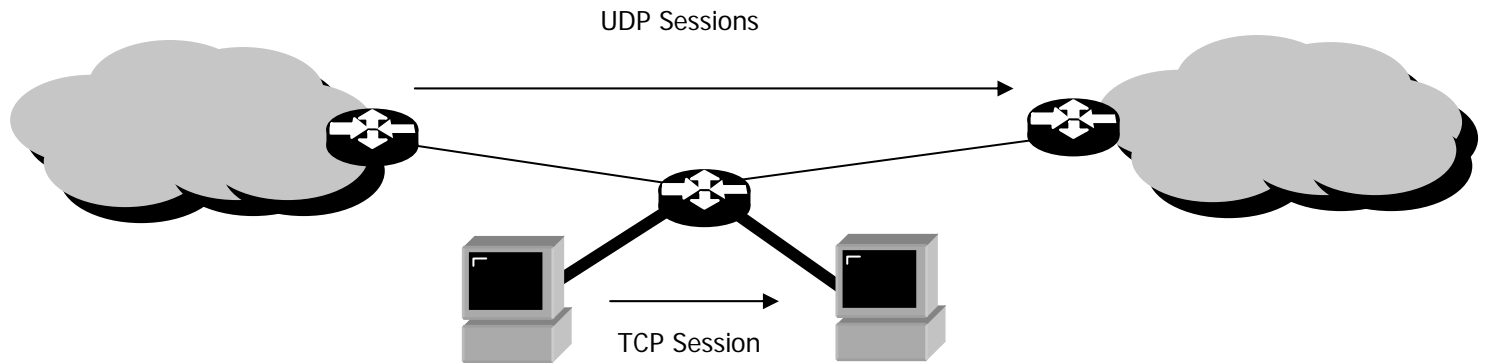
# The Multi-Service Problem

- Real-Time flows require:
    - short queues
    - admission control
    - priority queuing
  - Congestion-Managed flows require:
    - large queues
    - no admission control
    - explicit congestion notification
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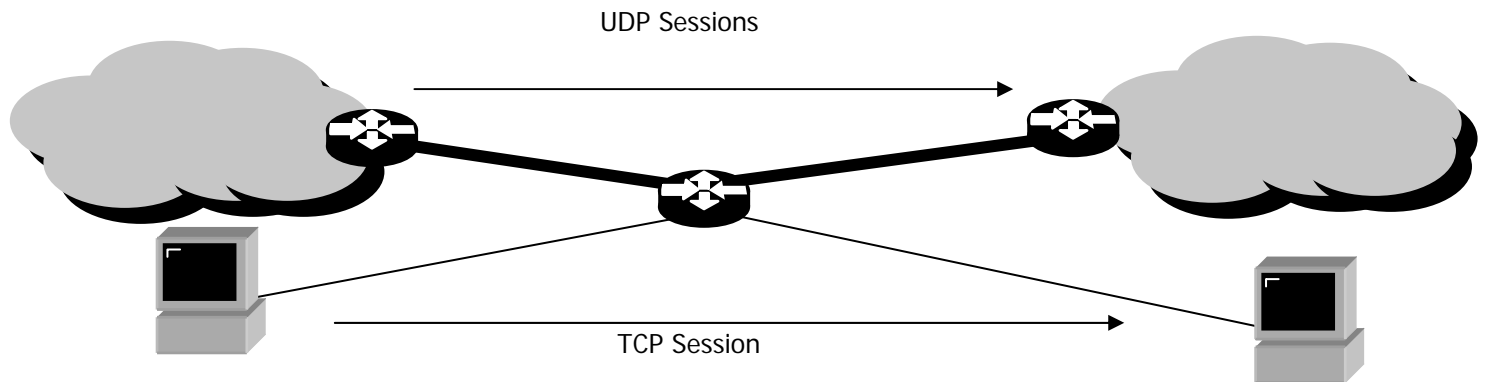
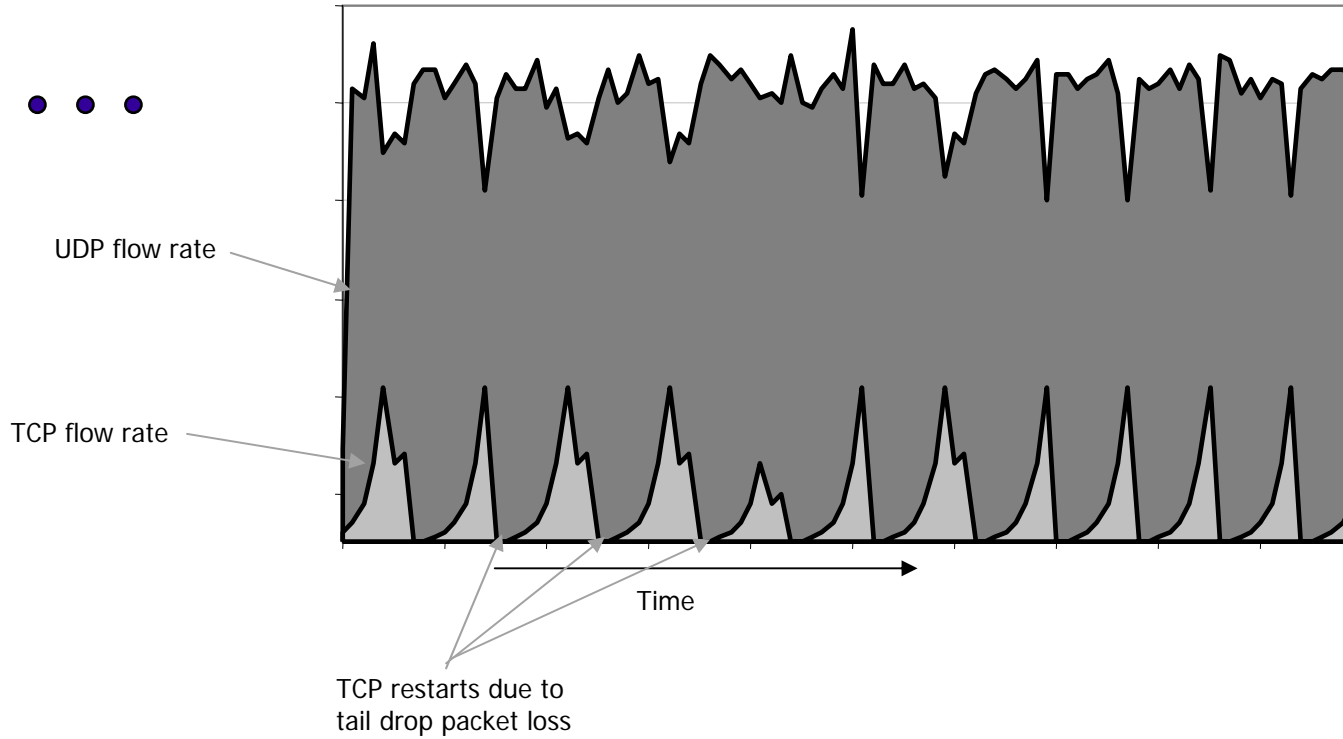
# Mixing TCP and UDP



Buffer starvation period  
as a result of a TCP burst




# Mixing TCP and UDP





# One Network Platform

- Can you mix Voice and Data at the packet level?
  - Voice over IP works - as long as:
    - small proportion of total traffic
    - queue lengths are kept short
    - some network inefficiency is tolerated
  - i.e. as long as the proportion of VOIP traffic is low compared to rate-adaptive traffic and the network is generally unloaded
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# The Multi-Service Network

- Does high quality service require resource reservations?
    - Can resource reservation be provided?
    - Is the cost of simulating time switching in a packet switched network higher or lower than the cost of operating a distinct time-switched network?
    - Where is the cross-over point?
    - Is service convergence and the mother-ship single platform operational model just a perverse throwback fantasy?
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