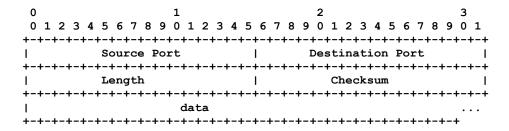
CS314 2011 27

UDP: User Datagram Protocol, Other Transports, Sockets

- IP is an unreliable datagram protocol
 - congestion or transmission errors cause lost packets
 - multiple routes may lead to out-of-order delivery
- UDP delivers exactly this service to user programs
- If senders
 - send too fast, routers or receivers cannot keep up (making congestion worse);
 - compete, capacity must be fairly shared
- UDP cannot solve these problems in any way

UDP header



Protocol Number or Next Header is 17 (decimal)

2

UDP header fields

- Port numbers used to find TCBs at each end
 - Note that source port is optional. Since there is no concept of a connection in UDP, it may not be needed
- Length
 - length in bytes of UDP header plus data
 - ill-advised to exceed the available MTU
- Checksum
 - 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero bytes at the end (if necessary) to make a multiple of two bytes
 - Pseudo-header is the same as for TCP

A day in the life of a UDP packet

- User A: Listen (portA)
- User B: Send (AddressA, portA, DataB)
- User A: Receive (DataB)
- User A: Listen (portA)
- That's if the packet gets through the network. If it happens to be discarded due to congestion or error, we get:
 - User A: Listen (portA)
 - User B: Send (AddressA, portA, DataB)
 - ... and the rest is silence

Why is UDP useful?

- Because UDP offers no error recovery and no error notification, it may appear useless
- In fact, on a network less than 100% busy, UDP packets usually get delivered. But a UDP-based application must include its own timeouts and error recovery. Mostly, it's easier to use TCP instead
- · Important UDP applications include
 - DHCP
 - DNS
 - RIP
 - SNMP (simple network management protocol)

which can all survive lost packets. Each listens on a well-known port number

References for UDP

- A few words in Shay 11.4
- Any of the TCP/IP books listed for IPv4
- RFCs:
 - RFC 768, the original definition
 - RFC 2460 (IPv6) modifies UDP's checksum formula

5

Other transport protocols

- RTP Real time Transmission Protocol
- SCTP Stream Control Transmission Protocol
- DCCP Datagram Congestion Control Protocol
- And related: ECN Explicit Congestion Notification

RTP - mainly for audio/video streams

- RTP data packets run over UDP on an even-numbered port
 - normally a port number above 16384
 - RTSP (RT Streaming Protocol) is layered on top of RTP
- RTCP (RTP Control Protocol) runs over UDP on the nexthigher (odd-numbered) port
- · RTP provides
 - Payload-type identification
 - Sequence numbering
 - Time stamping for synchronisation and jitter management
 - Delivery monitoring
- But with the unreliability issues of UDP
 - Video or audio codecs must allow for this

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SCTP

Background slide

DCCP

- A reliable, congestion-friendly protocol that has learned much from TCP
- Main differences:
 - Both ends can have multiple IP addresses, and the SCTP connection can switch between addresses (for example, in case of a routing failure for one of the addresses)
 - SCTP supports multi-streaming, i.e. separate virtual connections within the main SCTP connection
- Intended use was reliable connectivity for telephony signalling over the Internet
 - SCTP is quite general in applicability
 - Quite new and not widely used yet

•	DCCP behaves like a halfway house between UDP
	and TCP

- TCP's reliability and in-order delivery features introduce delays that are not OK for audio/video
- UDP's lack of congestion management causes network saturation when demand exceeds capacity
- DCCP establishes a connection (like TCP) and reports packet delivery (unlike UDP). It does not retransmit on error or attempt in-order delivery (like UDP and unlike TCP)
- DCCP offers two congestion management approaches
- Also makes use of ECN (next topic)
- Quite new and not widely used yet

9

10

Background slide

ECN

- Makes use of bits 6 and 7 in the IPv6 Traffic Class field or the IPv4 Differentiated Services field
 - 00 ECN not in use
 - 01 unused
 - 10 ECT flag
 - 11 CE flag
- ECT means "sender is ECN-capable"
- CE means "router is congested" and is interpreted by the receiving transport protocol
- A transport protocol that supports ECN will invoke a "slow down" mechanism when it receives a CE flag
 - Quite new and not widely used yet

Other Transport References

- RTP Shay 11.4, RFC 3550
- RTSP RFC 2326
- SCTP RFC 4960
- DCCP RFC 4340, 4341, 4342
- ECN RFC 3168

Sockets

- All transport protocols need a mechanism for upper layer software to access the transport
- The general concept is a notional "socket" that the application plugs into, embodied as a Socket API
 - Originated as "Berkeley sockets" on 4.2BSD Unix
 - Standard API is defined as part of Posix standard
- API includes calls to resolve DNS names into IP addresses, open and close sockets, and send and receive data
 - Plus many socket options for various purposes

Socket API overview (not a programmer's guide)

 socket is a function that creates a new socket data structure and returns a handle for it.

• close (mySocket) gets rid of a socket

Finding addresses (1)

- gethostbyname () is a function that takes a DNS name as a string and returns a structure containing the corresponding IP address
- In other words, it invokes the DNS resolver and the whole DNS lookup process
- Works well in an IPv4 network with one IP address per DNS name
- Inadequate for an IPv4/IPv6 network with multiple addresses per name

Finding addresses (2)

- getaddrinfo() overcomes the shortcomings of gethostbyname() for IPv4/IPv6 coexistence
- Allows user to express IPv6 preference
- Note that it <u>doesn't</u> return "the" address. The code that calls getaddrinfo is supposed to choose from a set of addresses
 - by default, assume addresses are ordered by system preference
 - if the first address doesn't answer, try the next one...
- The user code is more complex than with gethostbyname, but results in more robust application behaviour when there is any kind of network problem

14

13

Finding addresses (3)

- An AF_INET6 socket can be used for IPv6 or IPv4
 - if a "mapped" address like ::FFFF:10.1.2.3 is used, the socket will automatically use IPv4
 - Shay 11.3 is wrong to suggest that routers recognise these addresses - it is the sending host that decides to use IPv4 when mapped addresses are used with a socket
- The socket option IPV6_V6ONLY will force a socket to work only in IPv6 mode

Getting ready to talk

- bind() assigns an address to a socket
- listen() asks a socket to listen for incoming connections (TCP) or datagrams (UDP)
- connect() launches a connection (TCP SYN/ACK), or connects a program to a local socket (UDP)
 - shutdown () disconnects (TCP FIN/ACK)
- accept() accepts an incoming connection request (TCP)

17

18

Talking

- send() and recv() calls
 - and variants, or use write() and read()
- Streaming mode for TCP no relationship between individual send/recv calls and individual TCP segments
 - Hence the PUSH option in TCP, to force data into the receiver
- Datagram mode for UDP one send/recv for each datagram, and lost datagrams are truly lost
 - checksum errors give error returns from recv ()

Securing the socket layer: Transport Layer Security (TLS)

- Protects TCP sessions
 - Earlier versions known as SSL
- Uses a handshake procedure to negotiate crypto algorithm
- Uses server's public key to securely negotiate keys for the session
 - server presents a certificate to the client, including its public key. The certificate is cryptographically signed by a trusted certificate authority using its public key
- Following these negotiations, no 3rd party can intercept or inject traffic

19

Socket References

- Shay 12.2 (TLS is in 7.5)
- See your favourite Unix book or http://www.rt.com/man/
- With IPv6: RFC 3493 (and 3542)
- The Wikipedia article on Berkeley sockets is pretty good (but doesn't use getaddrinfo)
- POSIX standard: IEEE Std. 1003.1-2004 Standard for Information Technology -- Portable Operating System Interface (POSIX) See http://www.unix.org/version3/ieee_std.html