

## Quick UDP Internet Connections Multiplexed Stream Transport over UDP

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### What is QUIC?

Effectively replaces TLS and TCP out from under SPDY (predecessor of HTTP/2.0)

Provides multiplexed in-order reliable stream transport (especially HTTP) over UDP

Protocol is pushed into application space (unlike TCP which is handled in kernel)

### **Overview**

Why aren't current protocols enough?
 e.g., SPDY multiplexes streams, doesn't it?

What could make QUIC valuable?
 What could make it better(?) than SPDY?

Status of efforts?

### Why is SPDY fast?

• It is all about *latency (time till response)* 

- SPDY multiplexes requests over one TCP connection
- SPDY compresses headers
- What is the problem?

### Why isn't SPDY Enough?

### SPDY runs over TCP

- Lose one SPDY packet: all the streams wait
   HOL blocking
- Lose one SPDY packet, bandwidth shrinks
   Sharded connections have an advantage!!!
- SPDY may be slow to connect
  - TCP connect may cost 1-Round-Trip-Time (RTT)
  - TLS connect costs at least another RTT
- TLS and TCP are slow to evolve
   More importantly, they are very slow to deploy

   (at both ends, and in middle boxes!)

### **QUIC Goals**

- 1. Deploy in today's internet
- 2. Low latency (connect, and responses)
  - a. It is ALL about the latency
- 3. Reliable-stream support (like SPDY)
  - a. Reduce Head Of Line (HOL) blocking due to packet loss
- 4. Better congestion avoidance than TCP
  - a. Iterate and experiment
- 5. Privacy and Security comparable to TLS
- 6. Mobile interface migration
- 7. Improve on quality of sliced bread

### **QUIC Success Criteria**

The Internet is faster and more pleasant to use

Two paths: a) QUIC makes headway reducing latency b) TCP and TLS steam ahead, and perhaps use techniques advocated for QUIC

Either way: The users will win.

### Field Data, Plus Application Needs Drive Development

### Client side data from Chrome

- Real users; Real user machines; Real cross traffic; Real ISPs
- Aggregate data, and perform A/B experiments
- Server side instrumentation evaluates application impact
  - User happiness drives everything

## Can we really Deploy a UDP Protocol in Today's Internet?

- UDP works for gamers and VOIP
  - They really care about latency
- 91-94% of users can make outbound UDP connections to Google
  - Tested for users that had TCP connectivity to Google
- UDP is plausible to build a transport in today's internet

• See NAT Unbinding data in supporting slides

### NAT Unbinding: How much idle until unbinding?

#### Probability of Server Packet Never Reaching Client 100.00% 75.00% 50.00% 25 00% 0.00% 40 80 120 200 240 280 0 160

Seconds of idle before Server attempts to reach client

## How does QUIC achieve 0-RTT Connection Cost?

- Speculate that the server's public key is unchanged since last contact
   Propose session encryption key in first packet
   Include GET request(s) immediately after
  - Upgrade to Perfect Forward Secrecy ASAP
- Similar speculative techniques tried/developed in TLS and TCP
  - See <u>crypto doc</u> for fancy details
  - See supporting slides for some highlights

Congestion Avoidance via Packet Pacing

QUIC has pluggable congestion avoidance

- TCP Cubic is baseline
- Working on Pacing \*plus\* TCP Cubic
- Working on bandwidth estimation to drive pacing

### QUIC monitors inter-packet spacing

- Monitors one-way packet transit times
- Spacing can be used to estimate bandwidth
- Pacing reduces packet loss

### **Does Packet Pacing really reduce Packet Loss?**

- Yes!!! Pacing seems to help a lot
- Experiments show notable loss when rapidly sending (unpaced) packets
- Example: Look at 21st rapidly sent packet
  - 8-13% lost when unpaced
  - 1% lost with pacing
- See supporting slides on "Relative Packet ACK probabilities" for some details

# How might QUIC connection survive a Mobile Network Change?

### • TCP relies on src/dest IP/port pairs

- Mobile client (changing network/IP) means broken TCP connection :-(
- Broken TCP connection means big reconnect latency

### • QUIC relies on a 64 bit GUID in all packets

- Client source IP is used only to respond to the mobile client
- ...and of course with QUIC, if we lose....
  - ...then fast 0-RTT reconnect is a fallback

How can a Forward Error Correction (FEC) Packet help?

- Trade increased bandwidth for decreased latency
- QUIC has packet level Error Correction
   Keep a running-XOR of (some) packets
   Send XOR as an Error Correction packet
- ...but is packet loss bursty?
   XOR Error Correction won't work if we have several consecutive losses!

## Will Error Correction Coding really help?

- Packet loss is not that bursty :-)
   Example:
- 20 packets with about 1200 Bytes each
   Retransmit needed 18% of the time
- 20 packets plus FEC Packet
  - $\circ~$  Retransmit needed 10% of the time

5% extra bandwidth ==> -8% retransmits
 See support slide on Retransmit Probabilities for 1200B payloads for experimental data

### Status of Efforts (11/2013)

- Currently landing, limping, and evolving
  - In Chrome and in some Google servers
  - Trying to work as well as TCP Cubic
  - FEC built in... but not turned on
  - 0-RTT works when same server is hit.
- Try prototype Chrome canary
  - about:settings Enable QUIC :-) (must restart)
  - about:net-internals to look at activity
- Try test-server in chromium codebase
- Longer road to Crypto PCI compliance for handling credit cards :-(

### How can I contribute?

News group: proto-quic@chromium.org https://groups.google.com/a/chromium.org/d/forum/proto-quic Contribute to Chromium source tree! Evolving wire spec tries to record state-ofthe-Chromium-tree for landed code ...but debugging often drives changes **Design document** has motivations and justifications FAQ For Geeks addresses some questions

## Backup Slide areas: Other Nice Stuff in QUIC Design

- Defend against "Optimistic ACK Attack"

   a. Defend against Amplification Attacks

   Handle header compression (like SPDY)

   a. ...despite out-of-order arrival of packet (context)!

   Support TCP Congestion Avoidance

   a. Baseline: prevents Internet Congestion Collapse
- 4. 0-RTT Server-side redirects
  - a. Hand off service to other server without an RTT,
     while getting good crypto in that new server's stream

### More Nice Crypto Things In QUIC

- 1. Encryption used for both UDP port 80 (HTTP) and for port 443 (HTTPS)
  - a. ... but port 80 does not authenticate server
- 2. Connections upgrade to Perfect Forward Secrecy asap
  - a. After about 1 RTT
- 3. Packet padding to make traffic analysis a tad harder
- 4. FIN (like) and ACK packets authenticateda. No 3rd party teardown

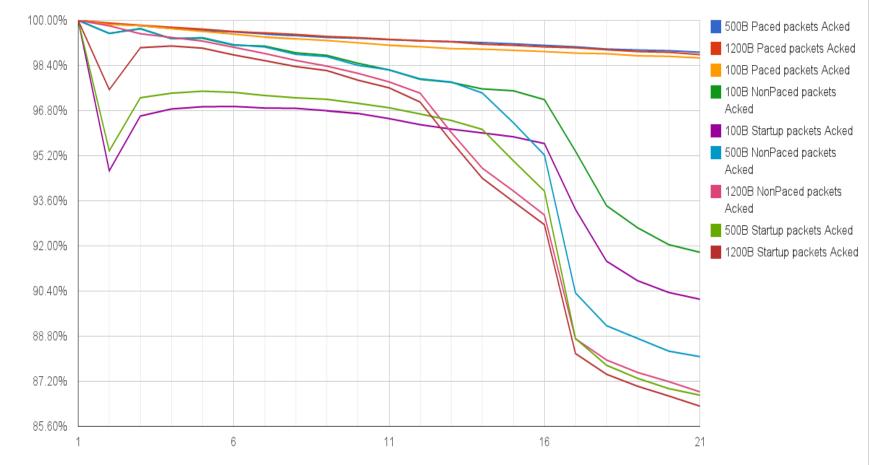
### **How is HOL Blocking Reduced?**

- UDP is not in-order, like TCP
   QUIC adds packet sequence numbers
- SSL crypto block depend(ed) on the previous block's decryption
  - QUIC uses packet sequence numbers as cryptoblock Initialization Vector (IV) source
  - QUIC collapses and reuses protocol layers!
- SSL encrypted blocks don't match IP packet boundaries :-(
  - QUIC aligns encryption blocks with IP packets
  - One lost QUIC packet won't stop the next packet from being decrypted :-)

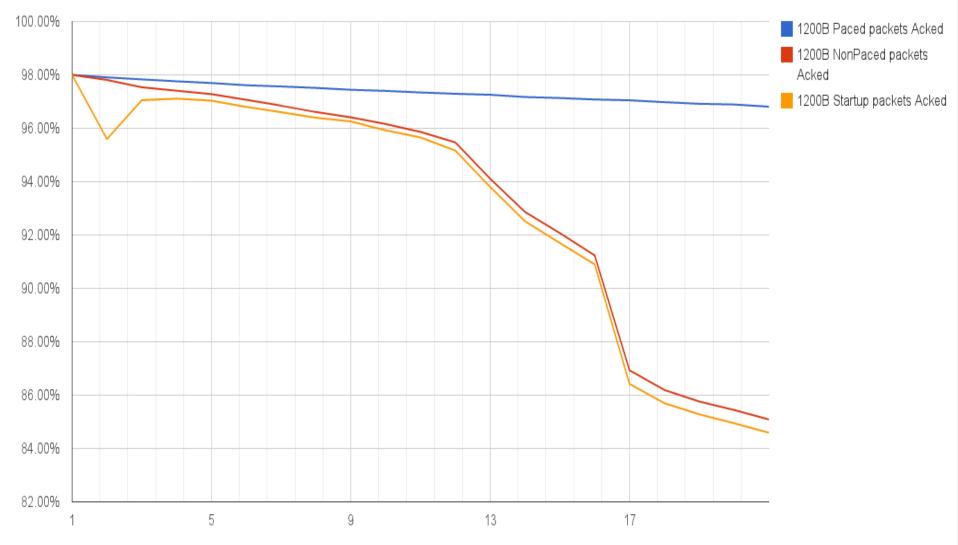
### Client->Server->Client round trip: 21 packets: 1200B vs 500B vs 200B

Relative Packet ACK Probability (relative to 1st packet probability)

Probability



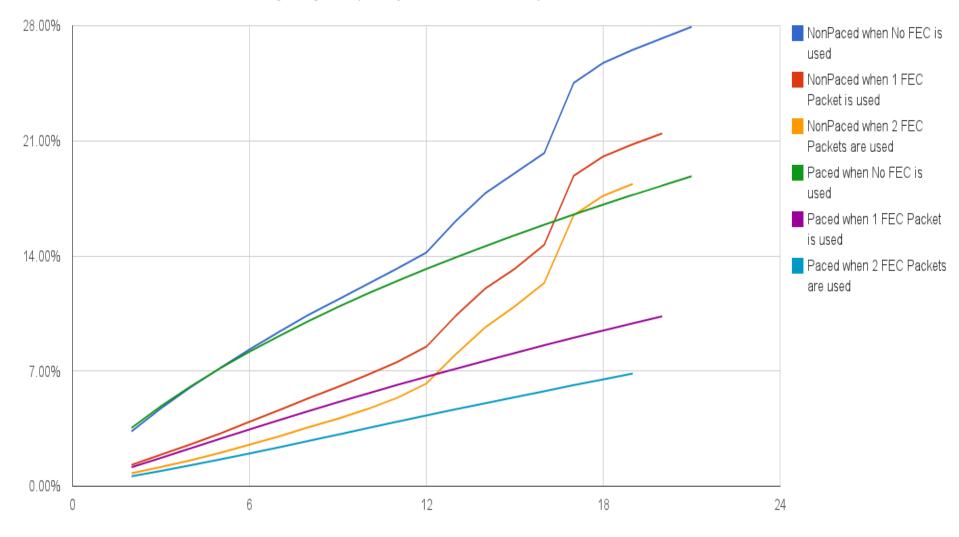
### **Round trip ACK Probabilities** Actual Packet ACK Probability



Packet Number

### **Retransmit-needed Probabilities**

#### Retransmit Probabilities for 1200 Byte Payloads (when port 6121 is not blocked)



### **NAT Unbinding Results Caveats**

- Did not correct for 1% ambient packet loss
   a. Could have sent N packets after pause
- 2. Did not validate internet connectivity
  - a. Users may have disconnected... so there is some disconnection conflation
- 3. Did not test to see if NAT was being used
  - a. IPv6 \*often\* avoids NAT
  - b. This could explain the 20% tail that "never"(?) unbinds