A Unified Approach for Repairing Packet Loss and Accelerating Channel Changes in Multicast IPTV

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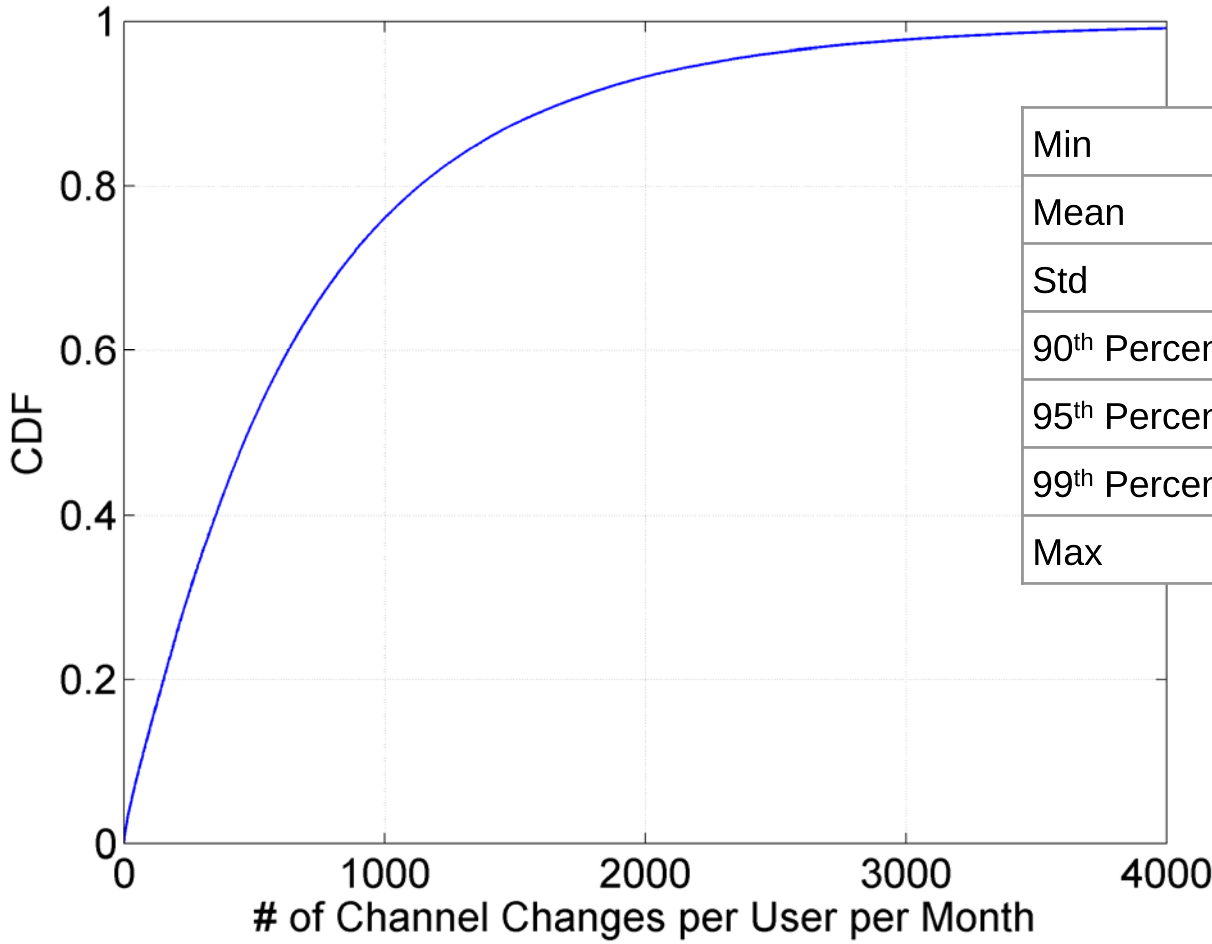
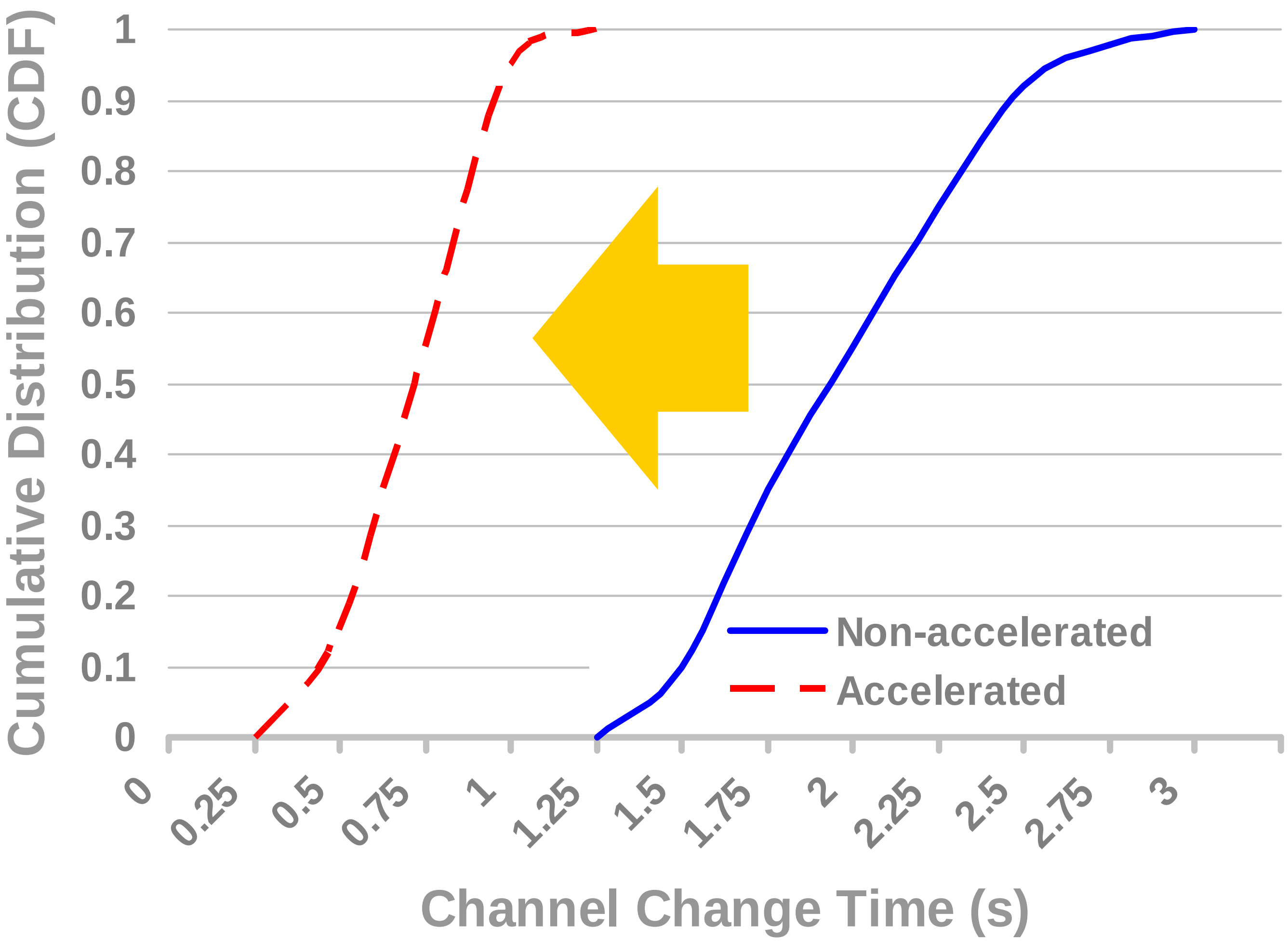
of Zappings per User per Month

Results are based on 227K+ users in NA



Min	1
Mean	726
Std	814
90th Percentile	1672
95th Percentile	2250
99th Percentile	3798
Max	24186

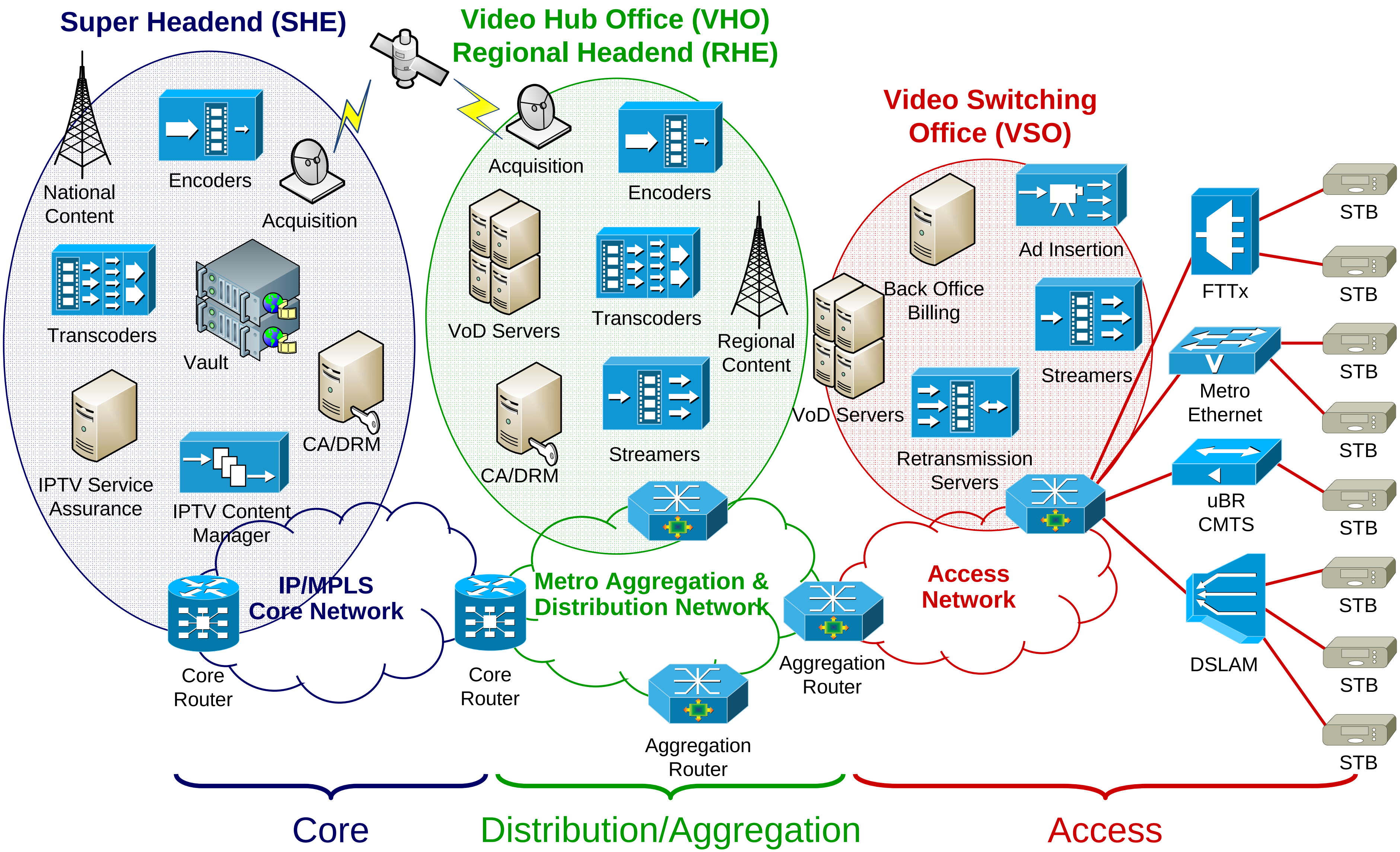
Goal: Consistent Short Zapping Times



Introduction

- **Zapping time is one of the top criteria an IPTV service is judged by**
  - Compression and encryption used in digital TV increase the zapping times
  - Multicasting in IPTV increases the zapping times
  - Zapping demand varies the zapping times
- **Service providers need a scalable solution that**
  - Is standards-based and interoperable with their infrastructure
  - Enables versatility and quick deployment
  - Keeps CAPEX and OPEX low
- **Our goals**
  - Provide constant, < 1s channel change times even in low-bandwidth networks
  - Benefit from existing protocol toolkits
- **In this study**
  - We overview IPTV transport issues and benefits of RTP
  - We discuss the channel changing problem in IPTV
  - We introduce the recent standardization efforts and present early results

End-to-End IPTV Network Architecture



Real-Time Transport Protocol (RTP)

- **Basics**
  - First specified by IETF in 1996, later updated in 2003 (RFC 3550)
  - Runs over any transport-layer protocol – UDP is much more widely used
- **Main Services**
  - Payload type identification
  - Sequence numbering
  - Timestamping
- **Extensions**
  - Basic RTP functionality uses a 12-byte header
  - RFC 5285 defines an RTP header extension mechanism
- **Control Plane – RTCP**
  - Provides minimal control and identification functionality
  - Enables a scalable monitoring functionality (Sender, receiver and extended reports)
- **RTP Transport**
  - Terrestrial, satellite and emerging IPTV networks dominantly use MPEG2-TS encapsulation
  - RFC 2250 defines a way to carry TS packets within RTP packets

Elements of Delay for Multicast Video

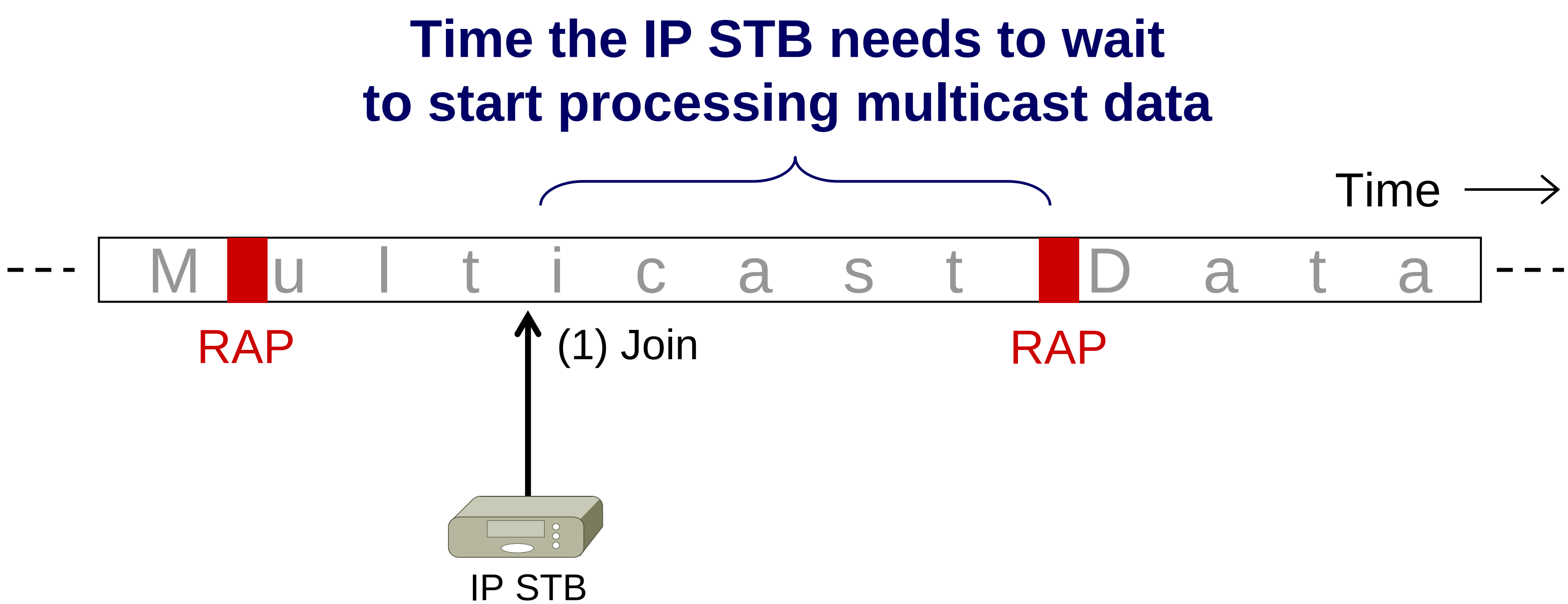
- **Multicast Switching Delay**
  - IGMP joins and leaves
  - Route establishment (Generally well-bounded)
- **Key Information Latency**
  - PSI (PAT/CAT/PMT) acquisition delay
  - CAS (ECM) delay
  - RAP acquisition delay
- **Buffering Delays**
  - **Loss-repair, de-jittering, application buffering**
  - **MPEG decoder buffering**

Key information latency and buffering delays are more critical in MPEG-based AV applications

Typical Zapping Times on DSL IPTV

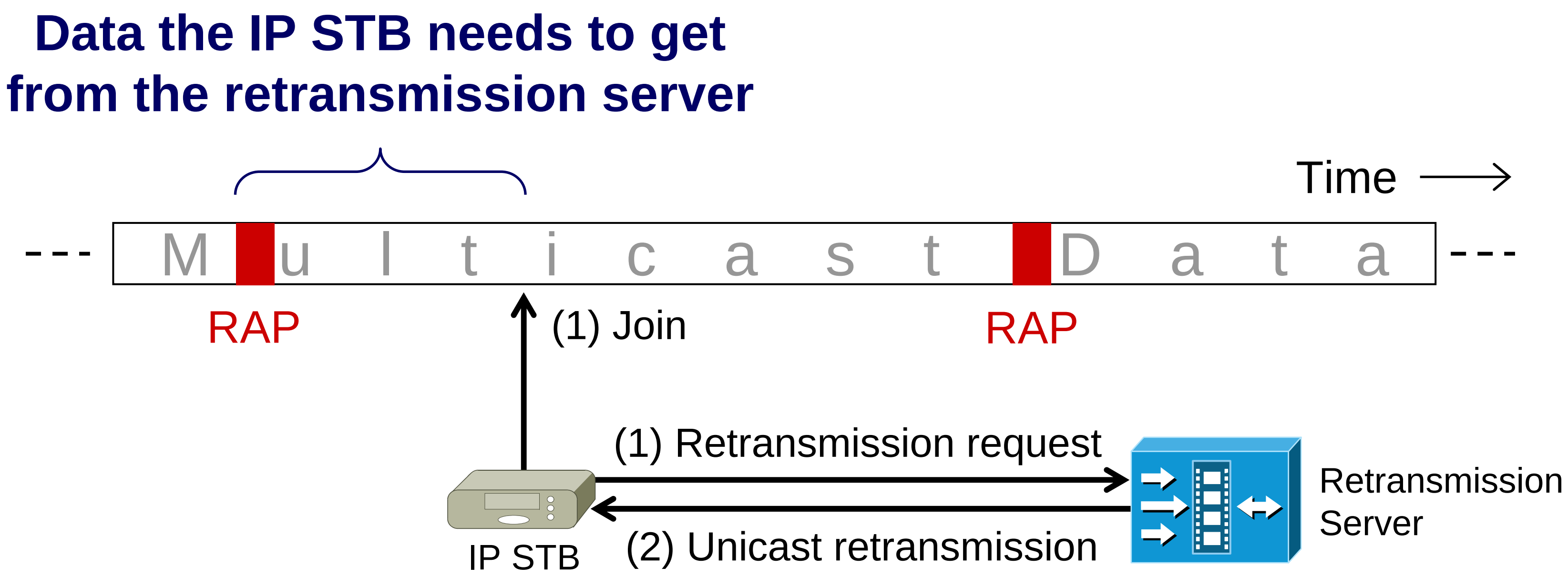
	Unit Time	Total Time
STB sends IGMP Leave	**< 10 ms**	
STB sends IGMP Join	**< 10 ms**	
DSLAM gets IGMP Leave	**< 10 ms**	
DSLAM gets IGMP Join	**< 10 ms**	**~ 20 ms**
DSLAM switches streams	**30 ms**	**~ 50 ms**
Latency on DSL line	**~ 10 ms**	**~ 60 ms**
STB receives PAT/PMT	**~ 125 ms**	**~ 185 ms**
Buffering		
De-jittering buffer	**< 50 ms**	**~ 200 ms**
Wait for CA	**< 50 ms**	**~ 250 ms**
Wait for I-frame	**0 – 3 s**	**0.2 – 3.2 s**
MPEG decoding buffer	**1 – 2 s**	**1.2 – 5.2 s**
Decoding	**< 50 ms**	**1.2 – 5.2 s**

A Typical Multicast Join



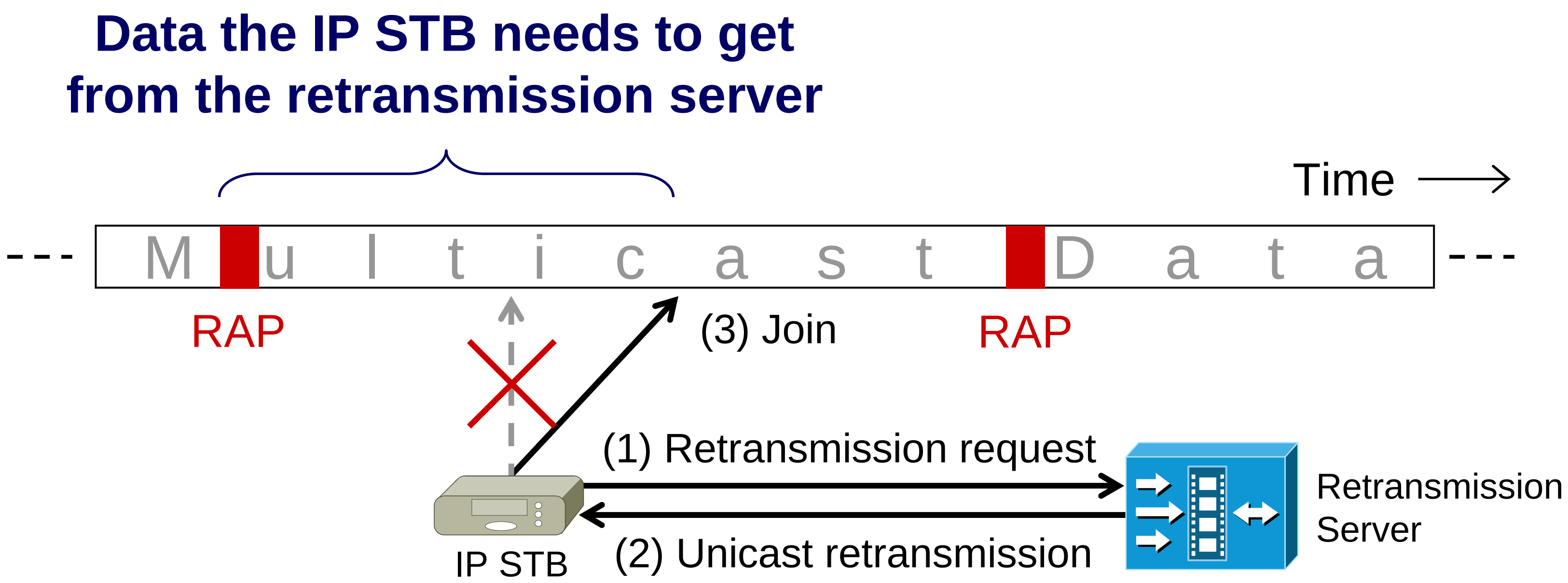
RAPs might be far away from each other
RAP data might be large in size and non-contiguous

Concurrent Multicast Join and Retransmission



If the residual bandwidth remaining from the multicast flow is small, retransmission may not be able to provide acceleration

Retransmission Followed by Multicast Join



More data are retransmitted due to deferred multicast join
However, IP STB ultimately achieves a faster synchronization

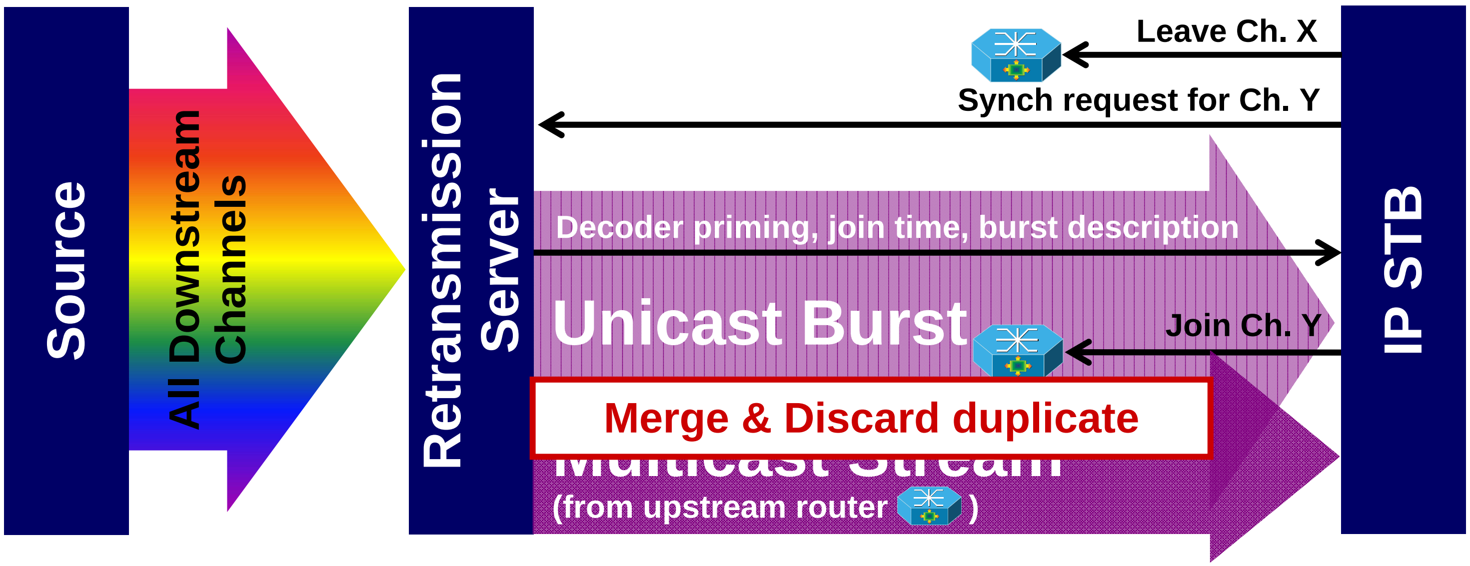
Proposed Solution

- IP STB says to the retransmission server:

"I have no synch with the stream. Send me a repair burst that will get me back on the track with the multicast session"

- Retransmission server may need to
  - Parse data from earlier in the stream than it is needed for retransmission
  - Burst faster than real time
  - Coordinate the time for multicast join and ending the burst
- This solution
  - Is applicable to any RTP-encapsulated multicast flow
  - Uses the existing toolkit for repairing packet losses in multicast sessions
    - RFC 3550 (RTP/RTCP)
    - RFC 4585 (RTP/AVPF)
    - RFC 4588 (RTP Retransmissions)
    - RTCP SSM (RTCP Extensions for SSM)

Rapid Synchronization

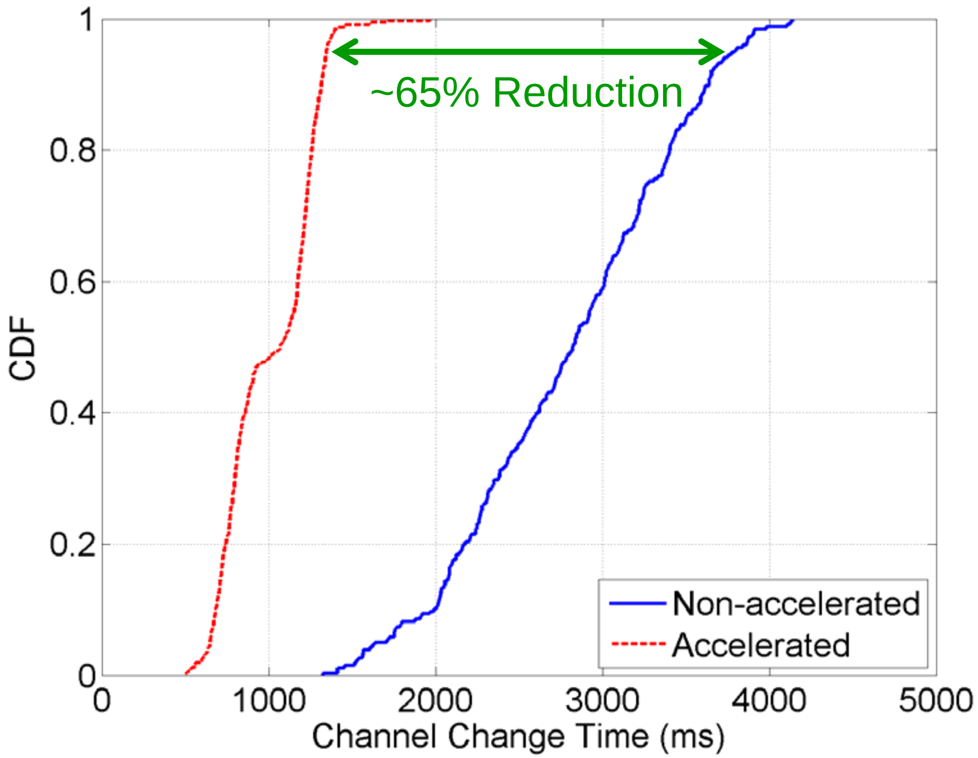


Retransmission server subscribes to all downstream multicast sessions

Experimental Setup

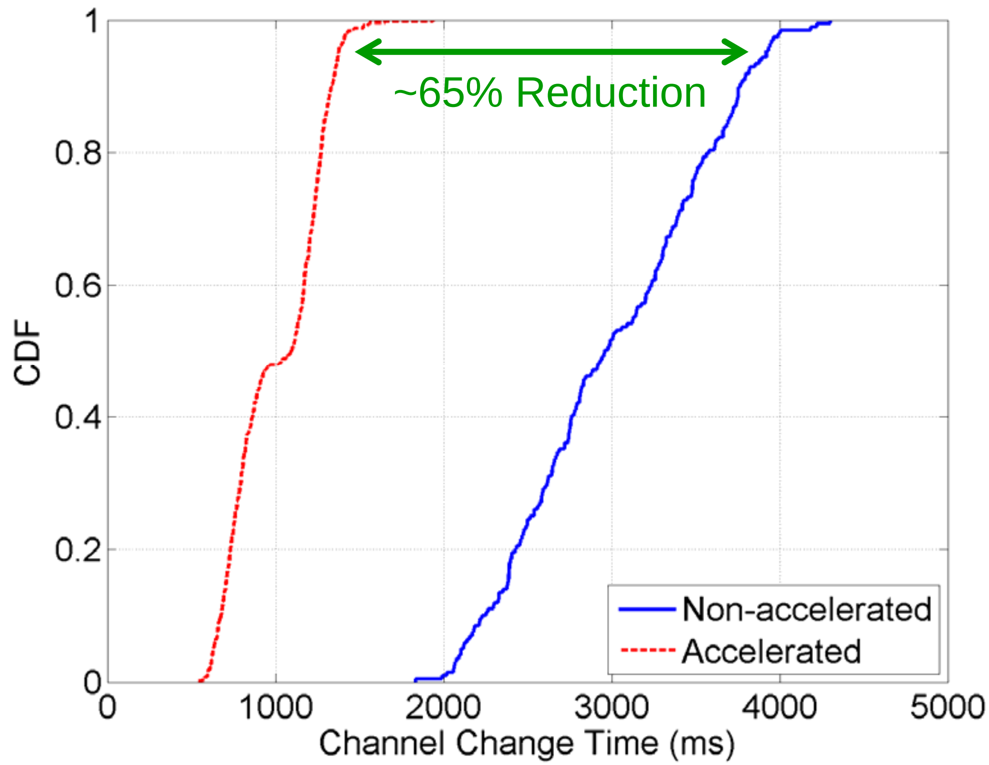
- **Comparison**
  - One IP STB with non-accelerated channel changes
  - One IP STB with accelerated channel changes
- **Video Streams**
  - High-detail, high-motion scenes of a movie
  - AVC encoded at 2 Mbps and 30 fps
    - One stream with 15 frames per GoP (Short-GoP)
    - One stream with 60 frames per GoP (Long-GoP)
- **Transport**
  - 1356-byte RTP packets
  - 20% additional bandwidth consumption for bursting
  - 500 ms loss-repair buffer in each IP STB

Short-GoP Results



	Min	Mean	Std	95^{th}	99^{th}	Max
Non-accelerated	**1323**	**2785**	**645**	**3788**	**4101**	**4140**
Accelerated	**501**	**1009**	**260**	**1345**	**1457**	**1965**

Long-GoP Results



	Min	Mean	Std	95^{th}	99^{th}	Max
Non-accelerated	**1831**	**3005**	**575**	**3920**	**4201**	**4300**
Accelerated	**536**	**1013**	**265**	**1377**	**1521**	**1937**

Open Source Implementation

- Client-side implementation is available as open source:

Documentation

http://tools.ietf.org/id/draft-versteeg-avt-rapid-synchronization-for-rtp

http://www.cisco.com/en/US/docs/video/cds/cda/vqe/3_0/user/guide/ch1_over.html

FTP Access

ftp://ftpeng.cisco.com/ftp/vqec/

Questions

The answer to life's problems aren't at the bottom of a bottle, they're on TV!

Homer Simpson

