NDN-RTC



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NDNComm 2014 Demo

- **Producer 1:** Live NDNComm HD streaming (1080p 30fps, 1.5Mbps)
- **Producer 2:** REMAP office webcam producer (SD, 30fps, 500Kbps)
- Demo 1:
 - Consumer for 3 streams: NDNComm, REMAP and Demo-2
 - Producer: webcam producer (SD, 25fps, 500Kbps)
- Demo 2:
 - Consumer for 3 streams: NDNComm, REMAP and Demo-1
 - Producer: webcam producer (SD, 25fps, 500Kbps)



NDN Real Time Conferencing Library

Goals:

- Real-time audio/video/text chat library which allows many-tomany conferencing over the NDN network and requires no direct communication between peers
- Starting point for NDN traffic congestion control algorithm research
- Test NDN-CPP library and NFD
- Traffic generator for the testbed

Initial gains over IP:

- No load on a publisher (network does content distribution)
- Intrinsic multicast (one-to-many and many-to-many scenarios)
- On track for peer-to-peer with no STUN, TURN, etc.

NDN-RTC library

- C++ code
- Linked against NDN-CPP and WebRTC libraries
- Interfaces:
 - Publish media (audio/video) streams
 - Fetch media (audio/video) streams from multiple producers
- Demo app is provided
 - Publishing audio/video stream
 - Fetching audio/video streams (multiple)



Publisher



Publisher. Multiple encoder threads



Publisher. Multiple media streams



Segmentation

- Encoded frames (1Mbps):
 - Key: ~30KB (20 segments)
 - Delta: ~1-6KB (~4 segments)
- Producer stores segments in app cache
 - Segment size 1000 bytes
 - NDN overhead ~330-450
 bytes
 - Complete segment less than MTU



User namespace



Consuming



Frame fetching



- Generation delay d_n^{gen} time interval between receiving an interest and satisfying it with data (*producer-side*)
- Assembling time d_n^{asm} time needed to fetch all frame segments (*consumer side*)
- **RTT**_n consumer-measured round trip time for the interest (*consumer side*)

Interest pipeline and retransmission

new frame - no segments fetched yet

frame being assembled (some segments fetched)

fully fetched frame



B₁ >= RTT, B₂ >= RTT Minimal buffer size >= 2*RTT milliseconds

Chase mode

- There is no direct coordination b/w consumers and producers
- Producer generates data at high rate (~20-30FPS) and this data becomes outdated fast
- Start-up time: consumer is aware that stale data is present in the network and tries to avoid playing it back
- Chasing mechanism:
 - Cache exhaustion:
 - Latest data can not arrive faster than it's being produced it arrives at producer's rate
 - Cached data arrives with the same frequency it was requested
 - Chase mode:
 - issue interest for the RIGHTMOST segment
 - upon receiving first segment start issuing interests for the next frames with interval d^{int} < Producer rate
 - Monitor *d*^{*arr*} frame inter-arrival interval:
 - If *d*^{arr} is increasing continue fetching
 - If *d*^{arr} is stable switch to "Fetch" mode

Chase mode (cont.)

darr



Future improvement (suggested by Dave Oran):

- 1. piggyback video sync data on audio stream
- 2. use audio stream for chasing instead of video

Forward Error Correction

- OpenFEC library
- Producer publishes parity data under separate namespace:
 <frame prefix>/<frame#>/parity/<segments>
- Consumer **may** additionally fetch parity data for enabling FEC
- If by the playback time frame is missing any segments FEC is applied as the "last resort"
- Amount of parity data is configurable (currently 20%)
- Collaborated with Daisuke Ando (Exchange student from Japan)
- Future improvement (suggested by Dave Oran): use framelevel parity data rather than segment-level

Demo app

- Console app
 - MacOS X 10.9 and up
 - Buildable from sources github.com/remap/ndnrtc
 - Redmine

redmine.nameddata.net/projects/ndnrtc

- Functionality:
 - Publish audio/video stream
 - Fetch multiple audio/video streams



Future steps

- Real-time Adaptive Rate Control:
 - In collaboration with Panasonic R&D department (Muramoto-san, Yoneda-san)
 - Keep low-latency transmission & best throughput
 - Maintain RTT fairness (self-fairness)
 - Consumer-driven
 - NW bandwidth estimation based on RTT and timeouts
 - Control interest rate according to bandwidth estimation
- Conference discovery (Zhehao Wang)
- Text chat (Zhehao Wang)
- Browser integration (Zhehao Wang)
- Security
- Desktop conference tool
 - Adding modularity to the existing code
- Compare to existing solutions
 - Can be RTC over NDN better than IP?
- Scalability tests

Areas for future research

- Interests pipelining
 - Express just enough interests to fetch needed frames and meet the deadline, but keep low latency
- Alternatives to cache exhaustion
 - How consumer can be sure that it's getting the latest data from the network without explicit producer-consumer signaling?
- Security
 - Trust model; signing and verification; encryption approach?
- Scalability
 - How many conference peers can there be?
 - What are the requirements for the forwarder?
 - What are the requirements for the peers?
- Relationship between forwarder strategy and application
 - Best route strategy 2

Links

- Source code
 - <u>https://github.com/remap/ndnrtc</u>
 - branches:
 - master current released version (v0.9.alpha4)
 - dev current development branch (v0.9.alpha5)
- MacOS binaries (library, demo, supporting files)
 - <u>https://github.com/peetonn/ndnrtc-archive</u>
 - Special branch for demo events:
 - demo/ndncomm2014
- Redmine
 - <u>http://redmine.named-data.net/projects/ndnrtc/issues</u>

Thanks

Q&A

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