

NDN-RTC: Real-time videoconferencing over Named Data Networking

Peter Gusev
UCLA REMAP
peter@remap.ucla.edu

Jeff Burke
UCLA REMAP
jburke@remap.ucla.edu

ABSTRACT

NDN-RTC is a real-time videoconferencing library that employs Named Data Networking (NDN), a proposed Future Internet Architecture. It was designed to provide an end-user experience similar to Skype or Google Hangouts, while taking advantage of the NDN architecture's name-based forwarding, data signatures, caching, and request aggregation. It demonstrates low-latency HD video communication over NDN, without direct producer-consumer coordination, which enables scaling to many consumers through the capacity of the network rather than the capacity of the producer. Internally, it employs widely used open source components, including the WebRTC library, VP9 codec, and OpenFEC for forward error correction. This paper presents the design, implementation in C++, and testing of NDN-RTC on the NDN testbed using a demonstration GUI conferencing application, NdnCon.

1. INTRODUCTION

Named Data Networking (NDN) is a proposed future Internet Architecture that shifts from the current host-centered paradigm of the IP Internet to data-centered communication. In NDN, every chunk of data has a hierarchical name, which can include human-readable components, and a cryptographic signature binding name, data, and the key of the publisher. Consumers of data issue “Interests” for these Data packets by name. Named, signed packets matching the Interest can be returned by any node on the network, including routers. NDN’s intrinsic caching can be leveraged by content distribution applications and significantly help to reduce the load on data publishers in multi-consumer scenarios. [5] Additionally, duplicate Interests for the same content can be aggregated in routers, further reducing the load on those publishers and the network. A full discussion of NDN is outside the scope of this paper; see the publications on the project website¹, including [?].

¹<http://named-data.net>

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Efficient content distribution has long been a driver application for NDN research and the broader field of Information Centric Networking (ICN) as well. We briefly discuss prior work in this area, including our own, in Section 3. However, *low-latency* application like “real-time conferencing” presents some particular design and implementation challenges that have not been widely explored in publicly available prototypes or the NDN and ICN literature. For example, obtaining the “latest data” from a network with pervasive caching, without relying on direct consumer-producer communication (which impacts scaling potential), while trying to keep application-level latency low, is challenging.

The NDN-RTC library was created to explore this arena experimentally. It was designed, implemented, and evaluated to explore NDN’s potential for scalable low-latency audio/video conferencing and “real-time” traffic more generally. This paper presents the current design and initial evaluation. NDN-RTC provides basic functionality for publishing audio/video streams, and for fetching these streams with low latency. This can be leveraged by desktop or web applications, such as the NdnCon sample application, for establishing multi-party conferences. The NDN network’s caching and interest aggregation are leveraged without architectural modification, with the NDN-RTC library ensuring low-latency communication.

We are working towards the goal of using NDN-RTC in NDN project videoconferences and meetings. To be practically useful in project communications, we also needed reasonable CPU and bandwidth efficiency, echo cancellation, and modern video coding: as a result, NDN-RTC is a C++ library which is built on top of the widely used WebRTC library, incorporating its existing audio pipeline (including echo cancellation) and its video codec (VP9).

The remainder of this paper is organized as follows: Section 3 covers background and prior work. Section 2 lists main project goals. Section 4 describes architecture of the library, designed namespace, data structures and algorithms used. Section 5 discusses implementation details. Section 6 evaluates main outcomes. Section ?? addresses existing issues and future work. Finally, Section ?? provides a conclusion.

2. DESIGN OBJECTIVES

The NDN project team uses application-driven research to explore NDN’s affordances for modern applications and to refine the architecture itself. Though it is based on what we learned from the NDNVideo project [5], NDN-RTC is a clean slate design with new goals. The initial objectives of the

NDN-RTC project are to explore *low-latency* audio/video communication over NDN, and to provide a working multi-party conferencing application that can be used by NDN project team members across existing NDN testbeds. At the same time, we attempt to preserve network-supported scalability by avoiding direct consumer-producer communication (e.g., Interests that require new Data to be generated by the producer for each request).

Over IP, low-latency audio/video conferencing applications typically establish direct peer-to-peer communication channels, for the best user experience. However, they face implementation challenges and inefficiencies related to the connection-based approach currently used in most IP conferencing solutions. Existing solutions scale poorly to high numbers of producers and consumers without dedicated aggregation units, for example.

Our high-level motivation for NDN-RTC is to see if “real-time” content publishing can be achieved like most other NDN data dissemination applications: The publisher 1) acquires and transforms data, 2) names, packetizes, and signs it, 3) passes it to an internal or external component that responds to Interests received from the “black box” of the NDN network that are matched against available named, signed, data chunks. The consumer issues Interests using appropriate names and selectors to the “black box”, at the rate necessary to achieve its objectives—as informed by performance of the network in response to its requests—and reassembles and renders them. Once the namespace is defined, the publishing problem, so far, seems straightforward. Complexity is at the consumer, which must determine what names to issue at what rate, to get the best quality of experience for the application. In the real-time conferencing arena, this means low latency access to the freshest data that the “black box” of the NDN network can deliver. Bidirectional communication is achieved by having each node participate as a publisher and consumer; conference setup and multi-party chat could be handled by applying techniques such as those developed in ChronoChat [?]; and multiple bitrates for consumers to use in adaptation are handled by simply publishing in multiple namespaces corresponding to multiple bitrates.²

Based on this high-level motivation, we developed several specific design goals:

- **Low-latency audio/video communication.** Library should be capable of maintaining low-latency (150-300ms) communication for audio and video, similar to driver applications such as Skype, WebEX and Google Hangouts.
- **Multi-party conferencing.** Publishing and fetching several media streams simultaneously should not require significant computational resources from the user and should maintain the same latency as in one-to-one conferencing.
- **Passive consumer & cacheability.** There should be no explicit negotiation or any coordination between active conference members as this may limit scalability and flexibility of use. Data should be cacheable for multiple consumers capable of decrypting it.
- **Data verification.** Library should provide content verification using existing NDN signature capabilities.
- **Encryption-based access control.** (Not implemented in this version.)

²Adaptive solutions based on scalable coding present some additional challenges that are left as future work.

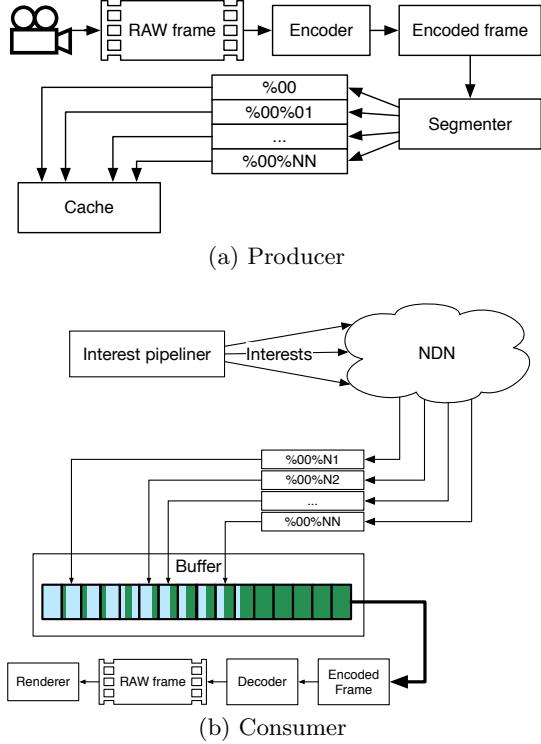


Figure 1: NDN-RTC producer and consumer operation.

3. BACKGROUND AND PRIOR WORK

To the authors’ knowledge, NDN-RTC is one of the few applications with perceptual real-time requirements being tested on an Information Centric Networking platform over a multi-hop testbed.

A non-real time video-streaming software solution, NDNVideo [5], was successfully tested and deployed on NDN, and proved high scalability. The project focused on developing random-access and live video for location-based and mixed reality applications. Another conferencing application (audio-only, however) was developed in [?]. It leveraged use of Mumble VoIP software, but used NDN as a transport. Initial effort for conference and user discovery was made in the recent work as well, suggesting that building on an existing, resilient platform is the best way to generate a usable application. Therefore, NDN-RTC was chosen to be built on top of WebRTC library, in order to utilize its’ audio-processing capabilities and video codecs, and potentially give an opportunity for easier integration with supported web-browsers.

4. APPLICATION ARCHITECTURE

There are two main roles defined in NDN-RTC: producer and consumer. With NDN, the paradigm of real-time communication shifts from push-based (when the producer writes data to the socket, and the consumer reads it as fast as possible) to pull-based (the producer publishes data on the network at own pace, while the consumer has to request data needed and manage incoming data segments).

Figure ?? presents a top-level overview of how NDN-RTC works. Local media capture and cache belong to the pro-

ducer. Media is stored in the cache which provides access to the data for all incoming interests. Remote playback represents the consumer: issues interests, prepares received media (assembles video frames from segments and re-orders them) and plays it back.

4.1 Producer

The producer's main tasks are to acquire video and audio data from media inputs, encode them, pack them into network packets, and store them in the cache for incoming Interests. In this way, complexity shifts to the consumer, and scaling is supported by the network.

In the case of video streaming, the producer uses video encoding in order to reduce size of the frames. There are two types of encoded frames: *Key* and *Delta*. Key frames contain most of the video information, and do not depend on any previous frames to be decoded. Whereas Delta frames are dependent on the previous frames (received after the last Key frame), and cannot be decoded without significant visual artifacts if any of the Key frames are missing.

Encoded frames vary in size, but average bitrate stays the same. For example, the average sizes of frames for 1000 kbps stream using VP8/VP9: Key frames are \approx 30KB, and delta frames are \approx 3-7KB. Therefore, depending on the underlying Internet Protocol used (IP in the existing NDN testbed), the producer may need to segment encoded frames into smaller chunks and provide clear naming conventions for the consumer to fetch them.

4.2 Namespace

As there is no direct consumer-producer communication, it is the producer's job to provide as much supportive information as possible, so that the consumer is able to use this information. These kinds should be reflected in the namespace: media data (segmented video frames and bundled audio samples), error correction data, and metadata.

4.2.1 Media

The NDN-RTC producer uses a *media stream* concept for describing published media. Media stream represents a flow of raw media packets (video frames or audio samples) coming from an input device. Streams usually derive names from their input devices. It is quite natural for a producer to publish several media streams simultaneously, if there is more than one device from which to publish media (for instance, video from camera, audio from microphone, and another video stream for computer screenshots). As all raw data should be encoded, the next level in the name hierarchy represents different encoder instances called *media threads*. Thus, media threads allow the producer to provide the same media stream in several copies (for instance in low, medium and high quality, so that the consumer can choose media thread more suitable for current network conditions).

NDN-RTC does not use any specific media container format for delivering media to consumers. Instead, encoded media packets are segmented if needed and published under distinctive hierarchical names. Video frames are separated into two domains as per frame type, *delta* and *key*, and numbered sequentially and independently. Sequence numbers for both delta and key frames start from 0. Next level specializes data type, either media data or parity. Parity data, if the producer opts to publish it, can be used by a consumer to recover frames that miss one or more segments.

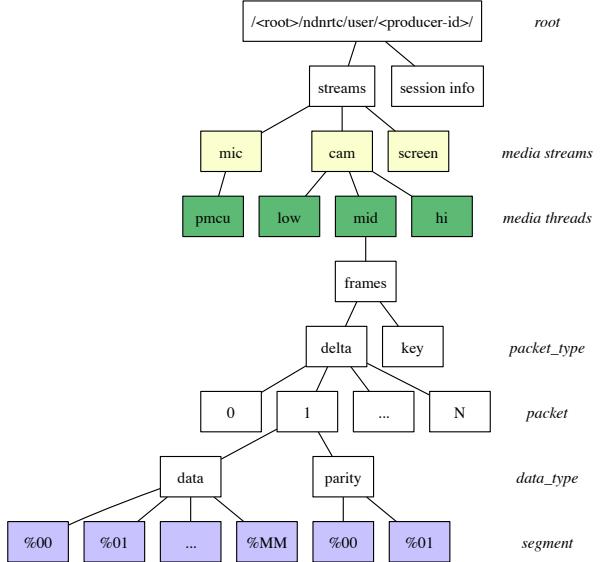


Figure 2: NDN-RTC namespace

The topmost level of the namespace defines individual data segments. These segments are also numbered sequentially, and segment numbers conform to NDN naming conventions [?].

In the case of audio streams, there are some differences. First, there are no key frames; therefore all audio packets are published under the *delta* namespace. Second, audio samples are significantly smaller than video frames and do not require to be segmented. In practice, it appears that multiple audio samples can be bundled into one data segment. Instead of segmenting audio packets, they are bundled together until the size of one data segment is reached, and published only after that.

Consumers need to know the producer's streams structure in order to fetch data successfully. In order to save consumers from traversing the producer's namespace, which can be time-consuming and unreliable, the producer publishes meta information about current streams under *session info* component. Thus, consumers can retrieve up-to-date information about the producer's state.

4.2.2 Metadata

Besides naming data objects, data names can be appended by some additional media-level meta information, which can be utilized by consumers regardless of which frame segment was received first. Three components are added at the end of every data segment name:

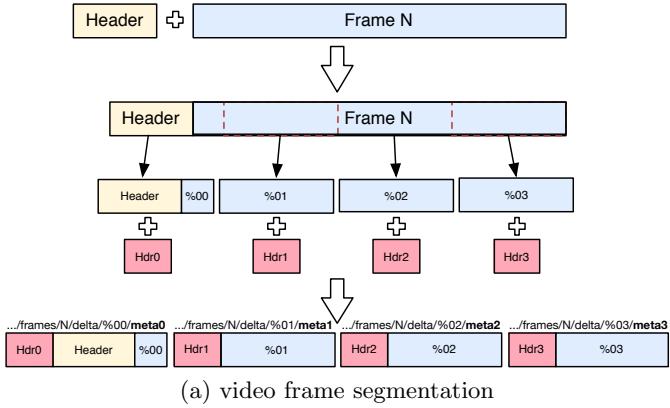
.../segment#/playback#/paired_seq#/num_parity

playback# - absolute playback number for current frame; this is different from the *frame#* which is a sequential number for the frame in its domain (i.e. Key or Delta);

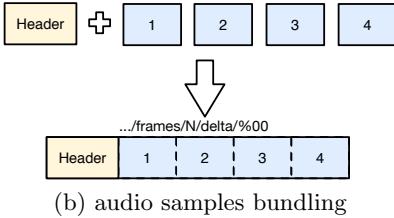
paired_seq# - sequential number of the corresponding frame from other domain (i.e., for delta frames, it is sequential number of the corresponding key frame which is required for decoding);

num_parity - number of parity segments for particular frame.

The pull-based nature of NDN and our experimental application's deliberate avoidance of explicit consumer-producer



(a) video frame segmentation



(b) audio samples bundling

Figure 3: Segmentation and bundling

synchronization (allowing publisher-independent scaling) have shown the importance of providing sufficient meta information on the producer side. Such information (which could include Interests' nonce values, Interests' arrival timestamps and data generation delays), if added to the returned data segment, may help consumers evaluate relevant network performance, detect congestion and assess whether incoming data is likely to be stale (delayed beyond the path delay). Furthermore, keeping historical data on the consumer side may help Interest pipelining in the future. For instance, providing the average number of segments per frame type helps consumers estimate the number of required initial Interests to fetch upcoming frames. This helps keep frame fetching cycles minimal.

4.3 Data objects

Producer generates signed data objects from input media streams and places them in cache instantly. Incoming Interests retrieve data from cache, if it is present, or forward further to the producer, if the requested data has yet to be produced. In such cases, the producer maintains a Pending Interests Table (PIT), which is checked every time a new data object is generated. If an Interest for a newly generated data object exists in the PIT, it gets answered, and PIT entry is erased.

4.3.1 Media stream

4.3.2 Metadata

Besides actual stream data, data objects contain some amount of meta information which is prepended during frames segmenting (see Figure 3(a)). There are two header types: *Frame header* and *Segment header*. Frame header is prepended to segment #0 of each individual video frame, and contains media-specific information (such as size of a frame), timestamp, current rate and unix timestamp, which can be

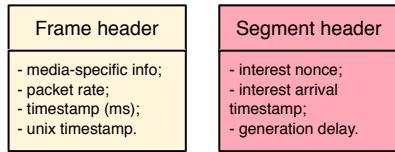


Figure 4: Frame and segment headers

used for calculating actual delay between NTP-synchronized producer and consumers (see Figure 4). Segment header is prepended to every segment of a frame. It carries network-level information which can be used by consumers to estimate current network conditions and origin of the data objects received:

Interest arrival timestamp: Timestamp of the interest arrival. Monitoring this value and interest expression timestamps over time may give consumers a clue about how it takes for Interests to reach the producer. This value is only valid when nonce value belongs to one of a given consumer's Interests.

Generation delay: Time interval in milliseconds between Interest arrival and segment publishing. A consumer can use this value in order to control the number of outstanding Interests. This value is only valid when nonce value belongs to one of the consumer's Interests.

Interest nonce:Nonce value of the interest which requested this particular segment. There are three meaningful cases: 1) *value belongs to the Interest issued previously* - consumer received non-cached data requested by Interest issued previously; 2) *value is non-zero, but does not belong to any of the previously issued Interests* - consumer received data requested by some other consumer; data may be cached; 3) *value is zero* - consumer received data which was cached on the producer side and never requested by anyone before.

Audio samples are not prepended by any segment header, however the whole audio bundle is prepended by the same frame header (see Figure 3(b)).

4.4 Consumer

The consumer implements more sophisticated algorithms to achieve the following goals:

- fetching the latest data available in the network;
- choosing appropriate media stream bandwidths provided by the producer (by monitoring network conditions);
- fetching media in correct order, for playback;
- handling network latency and packet drops.

The consumer takes into account that media packets are presented by separate segments in the network. Therefore, the consumer implements mechanisms of Interest pipelining and frame buffering (see Figure 1(b)). Interest pipeline issues Interests for individual segments, and is controlled by some higher-level logic which achieves one out of four consumer goals. It ensures that only the latest data is fetched. Packets re-ordering and drops and latency fluctuations absorption are attained by the frame buffer, which introduces buffering delay and re-arranges received frames in the playback order.

4.4.1 Frame fetching

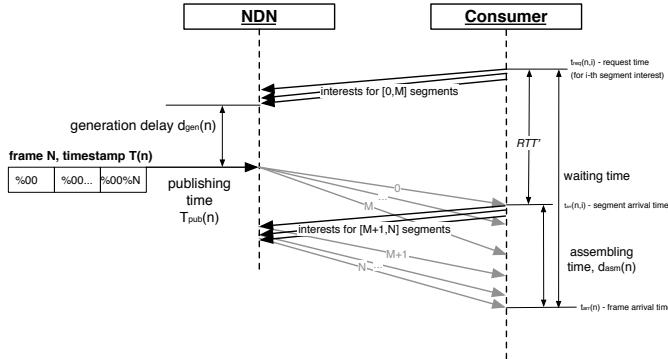


Figure 5: Fetching frame

The consumer does not know the total number of segments beforehand, unless the very first segment is fetched. In that case, the consumer can retrieve metadata from the received segment and acquire the total number of segments for the current frame. Therefore, at first attempt, the consumer makes a "best guess" at the number of segments needed to fetch, by issuing M Interests (see Figure 5). If Interests arrive too early, they will be added to the producer's PIT and stay there for the amount of time d_{gen} called **generation delay**. Once the encoded frame is segmented into N segments, these are published, and if $N \geq M$, Interests $0 - M$ are answered with the data which travels back to the consumer. Upon receiving the first data segment, the consumer determines the total number of segments for the current frame, and issues $N - M$ more Interests for the missing segments (if any). These segments will be satisfied by data with no generation delay (as the frame has been published already by the producer). The time interval between receiving the very first segment and when the frame is fully assembled is represented by d_{asm} and called **assembling time**.

Needless to say, additional round trips for requesting missing data segments increase overall frame assembling time and chances that the frame will be incomplete by the time it should be played back. This problem can be avoided if the consumer is able to make more accurate estimates of the number of initial Interests. Therefore, the following considerations were introduced in later versions of the library:

- The consumer should know what type of frame is to be fetched (as average number of segments varies greatly for different frame types).
- The consumer should track the average number of segments per frame type.

The first consideration was implemented by introducing separate namespaces for Key and Delta frames. The second consideration helps the consumer better estimate the number of initial Interests.

4.4.2 Buffering

As in traditional streaming applications, the consumer uses frame buffering in order to tackle out-of-order data arrivals and network delay deviations. Consumer-side jitter buffer is also used as a place for assembling frames by segments. However, the definition of "jitter buffer" is extended for NDN. In a traditional push-based approach, buffer can not contain empty frame slots; they are allocated/reserved

only when data arrives. A pull-based paradigm requires the consumer to request data explicitly. Therefore, after expressing an Interest, the consumers "knows" that new data is coming, and a frame slot should be reserved in the buffer. Practically, this means that there will always be some number of reserved empty slots in the buffer. Thus, jitter buffer's size has two measurements:

playback size - playback duration of all complete ordered frames by the moment;

estimated size = *playback size* + *number of reserved slots* $\times 1/\text{producer rate}$.

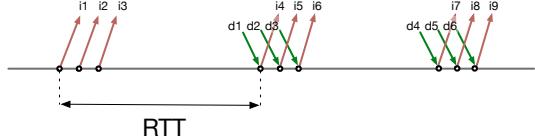
The difference between estimated size and playback size (RTT') cannot be smaller than the current average RTT value. In fact, keeping this value at a minimum indicates that consumer receives the most recent data with the minimal amount of outstanding Interests. Monitoring this value over time may help consumer get a better clue on the "sync" status with the producer. For example, consumer may use it during the fetching process, as will be discussed in the next section.

4.4.3 Interest expression control

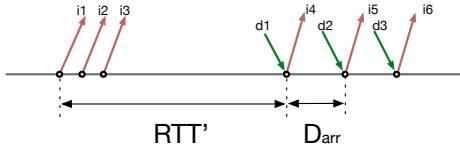
The key challenge in a consumer-driven model for video-conferencing is to *ensure the consumer gets the latest data in a caching network*, without resorting to direct producer-consumer communication that would limit scaling. To get fresh data, which can be cached but should not be the newest available for the consumer's path, the consumer cannot rely only on using such flags as *AnswerOriginKind* and *Right-MostChild*. The high frequency nature of streaming data makes no guarantees that the data satisfying those flags received by a consumer will be the most recent one. Instead, it is necessary to use other indicators to ensure that the network supplies up-to-date stream data. The basic solution is to leverage the known segment publishing rate and assume, under normal operation, that a series of old, cached samples can be retrieved more quickly than new data. The inter-arrival delays (D_{arr}) of the most recent samples follow the publishers' generation pattern, but cached data will follow the Interest expression temporal pattern. Therefore, by monitoring inter-arrival delays of consecutive media samples, consumers can make educated assumptions about data freshness (see Figure 6).

During bootstrapping, the consumer "chases" the producer and aims to exhaust network cache of historical (non-real time) segments. By increasing the number of outstanding Interests, the consumer "pulls cached data" out of the network, unless the freshest data begin to arrive. In order to control Interest expression, a concept of W (roughly an "Interest window") is introduced (see Figure 7). It expresses Interests only when $W > 0$. At every moment, W indicates how many outstanding Interests can be sent. Before the bootstrapping phase, the consumer initializes W with a value which reflects the consumer's estimate of how many Interests are needed in order to exhaust network cache and reach the most recent data.

W provides a simple mechanism to speed up or slow down Interests expression. Any increase in W value makes the consumer issue more Interests (Figure 7(b)), whereas any decrease in W holds the consumer back from sending any new Interests (Figure 7(c)). Larger values of W make the consumer reach a synchronized state with the producer more



(a) bursty cached data arrival, reflects interests expression pattern



(b) stable fresh data arrival, reflects publishing pattern

Figure 6: Getting the latest data: arrival patterns for the cached and most recent data

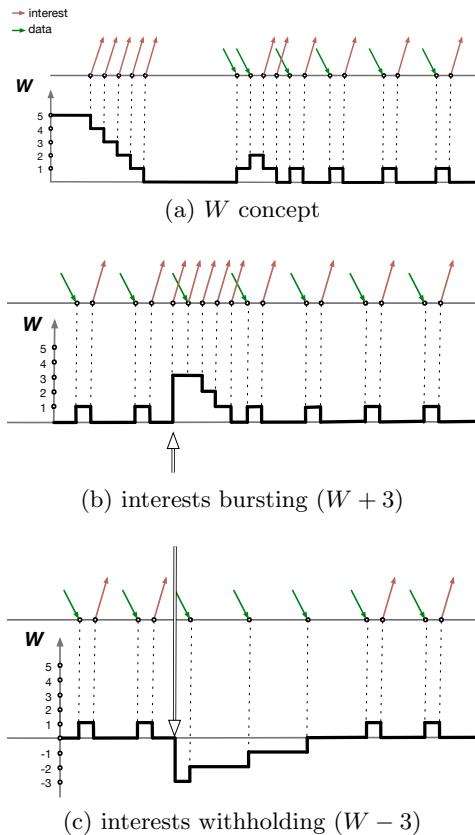


Figure 7: Managing interest expression

quickly. However, a larger value means a larger number of outstanding Interests and larger RTT' because of longer generation delays d_{gen} for each media sample. By adjusting the value of W and observing inter-arrival delays D_{arr} , the consumer can find minimal RTT' value while still getting non-cached data, thus achieving the best synchronization state with the producer.

For more complex scenarios of video streaming, the con-

sumer controls expression of ‘‘batches’’ of Interests rather than individual Interests, because video frames are composed of several segments. In this case, W is adjusted on a per-frame basis, rather than per-segment. In all other respects, the same logic (as above) can be applied.

The bootstrapping phase starts with issuing an Interest with the enabled *RightMostChild* selector, in Delta namespace for audio and Key namespace for video. The reason this process differs for video streams is that the consumer is not interested in fetching Delta frames without having corresponding Key frames for decoding. Once an initial data segment of a sample with number S_{seed} has been received, the consumer initializes W with initial value N and asks for the next sample data $S_{seed}+1$ in the appropriate namespace. Upon receiving the first segments of sample $S_{seed}+1$, the consumer initiates the fetching process (described above) for all namespaces (Delta and Key, if available). This bootstrapping phase stops when the consumer finds the minimal value of W which still allows receiving the most recent data, i.e., consumer reaches synchronized state with the producer and switches to a normal fetching phase where no adjustments for W are needed.

Application-level PIT. In most cases, consumers aim to express Interests for the data not yet produced, to be immediately satisfied when data is produced. The current NDN-CPP library provides a producer-side Memory Content Cache implementation into which data is published. However, this is only useful when data has been published and put in the cache before an Interest for this data has arrived. For the missing data, the Interest is forwarded to the producer application which stores it in the internal Pending Interests Table (PIT) unless requested data is ready. This functionality seems quite common for low-latency applications, and has now been incorporated into the NDN-CPP library implementation.

5. IMPLEMENTATION

NDN-RTC is implemented as a library written in C++, which is available at <https://github.com/remap/ndnrtc>. It provides publisher interface for publishing arbitrary number of media streams (audio or video), and consumer interface with a callback for rendering decoded video frames in a host application. OS X platform is currently supported; Linux build instructions will be added soon. The library distribution also comes with a simple console application which demonstrates the use of NDN-RTC library.

NDN-RTC exploits some functionality from several third-party libraries it is linked against with: NDN-CPP [6] is used for NDN connectivity. The WebRTC framework [2] is utilized in two ways: 1) incorporation of the existing video codec; 2) full incorporation of the existing WebRTC audio pipeline, including echo cancellation; 3) OpenFec [1] library is utilized for forward error correction support.

Apart from the library, first desktop NDN videoconferencing application NdnCon [3] was implemented atop NDN-RTC. It provides convenient UI for publishing and fetching media streams, text chat and organizing multi-party audio/video conferences.

6. EVALUATION

During the course of this project, there were several architecture design iterations that introduced successive im-

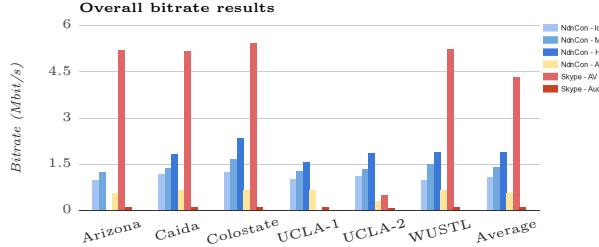


Figure 8: 2-peer conference tests compared to Skype

provements in the overall quality of the algorithms. (These are described further in this section.) Each iteration tackled problems that were revealed during tests (mostly in practice rather than simulated).

6.1 Video streaming performance

Video streaming performance in previous versions of NDN-RTC largely suffered from video "hiccups", even when being tested on trivial one-hop topologies. The cause of this problem turned out to be an inefficient frame fetching process.

As described in previous sections, Key frames are usually much heavier than Delta frames and require more data segments for delivery. In early library versions, this differentiation was not reflected in the producer's namespace, which made consumer to pipeline equal number of initial interests (M on Figure 5) for both frame types. This resulted in larger assembling times (d_{asm}) for the Key frames and additional round-trips of missing interests. Increased assembling time quite often caused skipping incomplete Key frames as they were not assembled by the time they should be played out. Eventually, all the consequent Delta frames were skipped as well which degraded overall video streaming experience by introducing video "hiccup" effect.

Having separate namespace for Key frames allowed consumer to maintain separate interest pipelines per frame type and collect historical data on the average number of interests required to fully retrieve one frame of each type in one round trip.

6.2 Quality of experience

A series of tests were taken in order to assess efficiency and quality of service compared to Skype calls. Each test was comprised of six runs of 2-person 5-minute conference talks using NdnCon (GUI conferencing application build atop NDN-RTC):

- 3 runs of audio+video with low, medium and high video bandwidths settings (0.5, 0.7 and 1.5 Mbit/s accordingly);
- 1 run of audio-only conference;
- 1 run of Skype audio+video conference;
- 1 run of Skype audio-only conference.

Tests were taken across existing NDN testbed between REMAP hub (aleph.ndn.ucla.edu) and six other hubs. Tests covered both one-hop and multi-hop (with several intermediate hubs) topologies.

Figure 8 shows overall bitrate usage results. Firstly, whereas Skype has fully utilized link capacity between peers and delivered higher bitrate videos, NdnCon did not adjust to the current network conditions which makes feature of adaptive

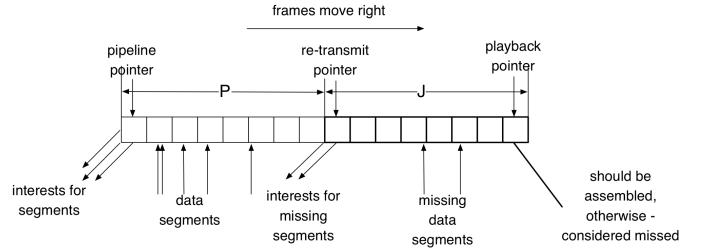


Figure 9: Bufferization in earlier library versions

rate control highly desirable.

Actual average bitrates are turned out to be slightly higher than pre-configured video streams – 0.7, 0.9 and 1.8 Mbit/s for low, medium and high bandwidths – which can be explained by NDN packet overhead which is about 280-330 bytes large and account for $\approx 30\%$ of segment size (1000 bytes). Having such large overhead makes transferring audio samples in separate segments highly inefficient. Thus, further improvement to the algorithms was to bundle consecutive audio samples unless they fill the size of a segment. With 90 Kbit/s audio, approximately 5 audio samples can be added to 1000-bytes data segment. This improvement eventually reduced audio bandwidth effectively (and interests number on consumer side) and made it comparable to Skype audio bandwidths.

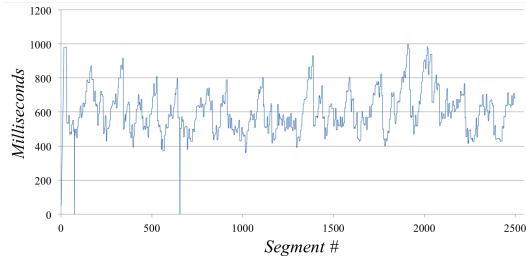
The overall user experience for these 2-person conferences was subjectively assessed as being higher than average - 3 points on a scale from 0 to 5 (Skype calls were taken as a 5-point user experiences).

6.3 Consumer-Producer synchronization

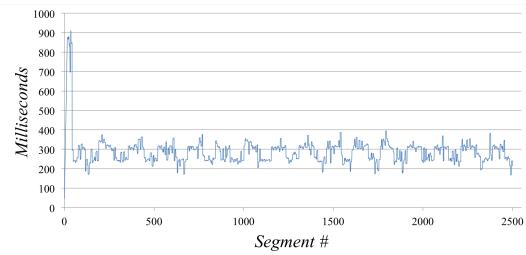
In previous library versions, the consumer "chased" the producer's time-series data by exhausting cached data and issuing large number of outstanding interests. However, there were no mechanism for consumer to figure out how early those interests are issued and whether interest expression should be postponed in order to eliminate timeouts. For two similar test runs (one-hop topology) the number of timed out interests and retransmissions could vary greatly (either $\approx 1\%$ or $\approx 50\%$). The latter case happened due to incorrect consumer's synchronization with the producer - interests were issued too early so that they were timed out before any data has been produced. This problem could be solved by increased interests' lifetime. However, for previous library versions, bufferization mechanism (see Figure 9) dictated interests' lifetime - in fact, in order to maintain re-transmission checkpoint, all interests entering the buffer should have lifetime equals half of the current buffer size. This approach resulted in unavoidable interests time outs in cases when consumer issued interests far too early before the actual data was produced.

For the current version of the library, re-transmission checkpoint is placed at RTT milliseconds from the end of the buffer ($J = RTT$ on the Figure 9), this, together with updated NFD retransmission strategy [4], allowed larger interests lifetimes.

Moreover, the problem described above could not have happened if consumer would know that it issues interests



(a) $d_{gen} \approx 600ms$ resulted in 50% retransmissions



(b) $d_{gen} \approx 310ms$ - no redundant retransmissions

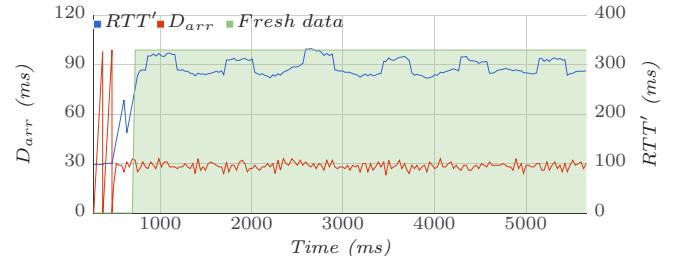
Figure 10: Two separate runs of earlier library version on similar topology (one-hop): random results for data generation delay d_{gen} due to poor consumer-producer synchronization

too early. Chasing algorithm in older library versions was exhausting network cache too aggressively - interests were issued constantly unless they fill up the buffer. After that, frames were retrieved at producer's rate and more interests issued unless consumer starts to receive the most recent data. This approach lacks from any knowledge about issued interests and data generation delay and RTT' are not taken into account.

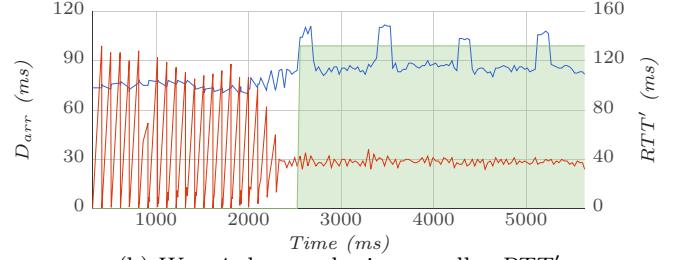
With introduction of W concept, consumer have full control of the interest expression. Figure 11 shows how bigger value of W helps to exhaust cache faster. The number of outstanding interests is controlled by a consumer and directly influences how fast consumer can "chase" the producer. Every time a new interest is expressed, W gets decremented and when new data arrives, W is incremented, thus allowing consumer to issue more interests. W concept allows "lazy" start for consumer - by specifying smaller W , consumer issues less interests. Further, consumer observes cache exhaustion by monitoring D_{arr} and, if cache has not been exhausted during allocated time, consumer may increase the value of W in order to express more interests. Similarly, consumer may opt to decrease W in cases where original value resulted in too aggressive behaviour.

6.4 Community use

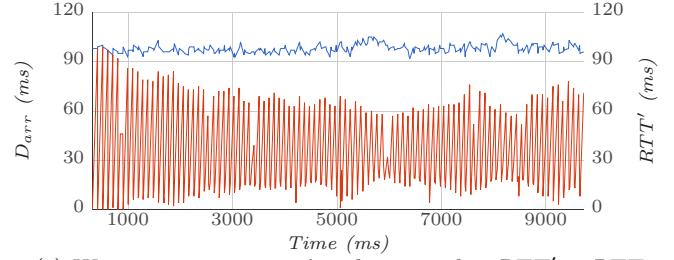
Initial attempts of deploying NdnCon conferencing application for NDN Community were made in early 2015. NdnCon was used to stream NDN seminar over the existing NDN testbed. To do that, an audio/video bridge was set up using third-party tools like Soundflower and CamTwist Studio allowing to publish captured screen and audio feed from the seminar (taken in Cisco WebEX conferencing tool) over the NDN. Figure 12(a) shows NDN testbed utilization during 1-hour conference call. It is expected that media



(a) $W = 10$: short chasing, larger RTT'



(b) $W = 4$: longer chasing, smaller RTT'



(c) $W = 3$: consumer can't exhaust cache, $RTT' = RTT$

Figure 11: Larger W decreases "chasing" phase, but increases RTT' for the same network configuration ($RTT \approx 100ms$)

streams were consumed by 5 to 8 people. Overall quality was subjectively satisfying as reported by users.

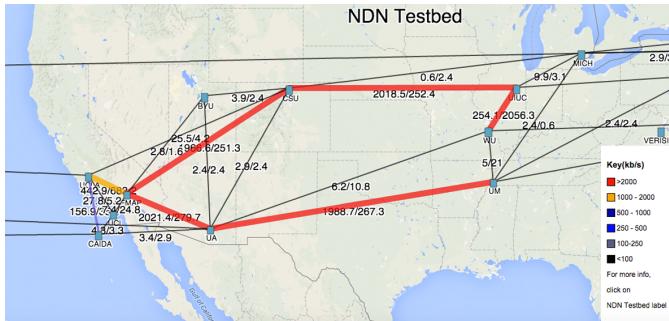
Another run was made in an attempt to test multi-party conferencing ability. This test included 4 peers each publishing 3 video streams and 1 audio stream and fetching 1 video and 1 audio stream from each of the other participants. Participants were distributed across four campuses - UCLA, REMAP, CAIDA and WashU (Figure 12(b)). These results were subjectively satisfying, however for one user (connected to WashU hub), audio cut-offs happened more often than for the other participants. The root causes of this have not been revealed yet. Many-to-many test scenarios are complex and will be taken for future work.

7. CONCLUSION AND FUTURE WORK

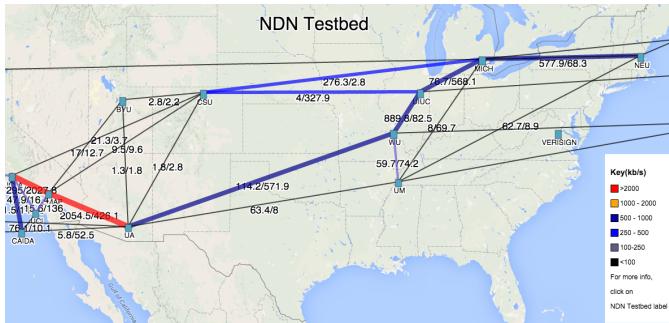
Discuss quality of experience.

Future work:

Adaptive rate control. In the current library design, the producer may deliberately choose to publish several copies of the same video stream with different encoding parameters, thus allowing consumer to select the most appropriate stream for current network conditions. However, the selection is made manually and depends on user's perceptual assessment of the retrieved media. Implementation of



(a) NDN testbed utilization during biweekly NDN seminar



(b) NDN testbed utilization during 4-peer call between UCLA REMAP, WashU and Caida hubs

Figure 12: NDN testbed utilization during one-to-many and many-to-many scenarios



Figure 13: NdnCon screenshot [@@REPLACE]

adaptive rate control would simplify this process, and allow the network to be utilized more efficiently based on current conditions.

Scalable video coding. An elegant way to offload the producer from publishing multiple copies of the same video stream in different bandwidths is to utilize Scalable Video Coding. By reflecting SVC layers in the namespace, consumer will have more freedom for adapting media streams to the current network. This opportunity should be explored and added in a future versions.

Audio prioritization. For quality of experience in typical audio/videoconferencing applications, audio should be prioritized over video. This can be done at the application level and we may provide such support in the future.

Encryption-based access control The current NDN-RTC design supports basic content signing and verification. However, further basic security features have yet to be implemented, e.g., media data encryption, consumer access control.

Conference management Work on ndncon. unknown but verified publishers trust;

8. ACKNOWLEDGEMENTS

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