WebRTC-based Premium Streaming Ecosystem

DASH-IF, 28 May 2021

Presenters

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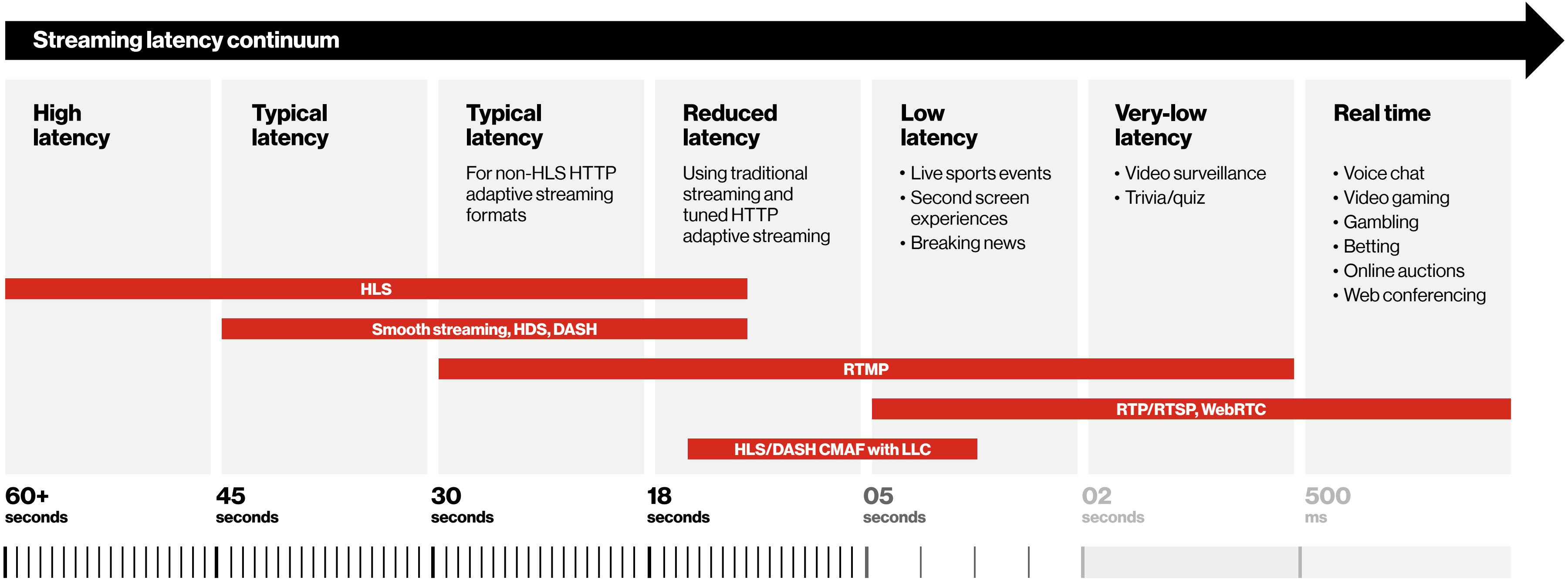
Agenda

1. Use Cases
2. Ecosystem & Gaps
3. Service Discovery & Negotiation
4. Other Ecosystem Topics
    a. DRM
    b. Ad Insertion
5. Next steps

Streaming Latency: It’s a continuum



Interactivity Requires Real-time

Examples of User Experiences

- Multi-angle user-selectable content, synchronized in real-time
- Conversations between hosts and viewers
- Co-watching of content with a group of friends (VOD & Live)
- Interactive news, where viewers add their own content to the story

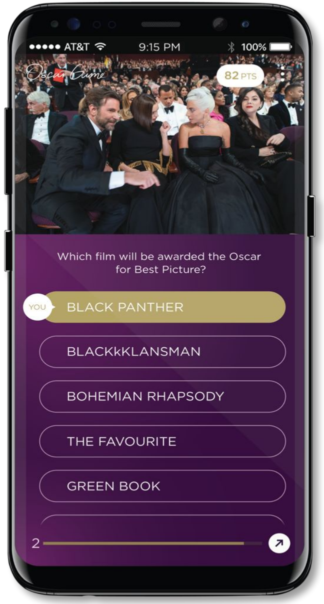
Interactivity

- host ⇔ viewers
- user ⇔ user

Industry Verticals

- Gambling
- Gamification
- Sports
- Trivia / Online Quiz Games
- Auctions
- Online shopping

Premium Video in Production Use Cases



Quality & Scale in Real-time with WebRTC

Cheltenham Festival

Achieved **478,000** peak concurrent viewers

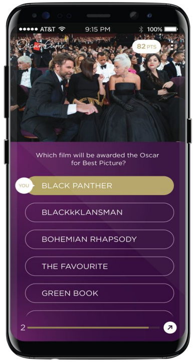
Grand National

10 races topped **100,000** peak concurrent users
Two topped **175,000** concurrent users
Two topped **225,000** peak concurrent users

Who Wants To Be A Millionaire

Mobile app companion
Over **100,000** peak concurrent players each week

The Oscars



Total views: **1.4M**

Peak Concurrent Users = **114,000**

Stream Join Rate = **80,000/sec**

Q12 Trivia Game

Delivered **220,000** concurrent users on iOS and Android

10 weeks of **200,000+** concurrent users

What is WebRTC?

- Originally, created for real-time communication for the Web
- W3C Standard
- Supports video, audio & data streaming
- Built into all modern browsers across desktop and mobile devices
- Increasingly used today for real time streaming of premium content

Premium Streaming Ecosystem

Acquire

- SMPTE SDI
- SRT SECURE RELIABLE TRANSPORT
- ZiXi
- IETF RTSP/RTP
- MPEG Transport Stream
- RTMP
- RIST RELIABLE INTERNET STREAM TRANSPORT
- NDI

Adapt

- H.264 MPEG-4/AVC
- opus
- AV1
- H.265 High Effecieny Video Coding
- VP8
- DOLBY
- DOLBY VISION • ATMOS
- VP9
- MPEG-H AUDIO
- dts
- HDR10+

Enhance

- SCTE
- n
- CEA
- DVB

Deliver

- WebRTC
- RTMP
- MBR
- mpeg-DASH
- HLS

Consume

- Mac
- iOS
- tv
- Windows
- XBOX
- android
- PlayStation
- Linux

Ecosystem Gaps

- How a viewer discovers and joins an experience
- Session Negotiation
- Captions / subtitles
- Timed Metadata
- Ad Insertion
- Digital Rights Management (DRM)
- Advanced Audio & Video Codecs

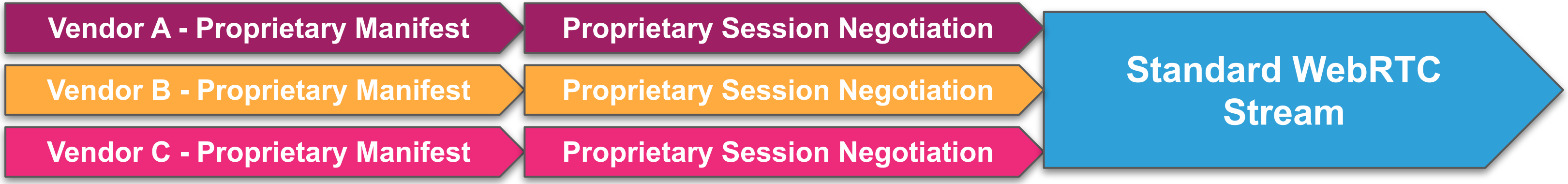
Bridging these gaps with standard solutions will maximize interoperability and streamline development and adoption

From Discovery to Streaming

Current and future states of how a viewer discovers and joins an experience

The manifest contains multiple ‘adaptation sets’ for camera angles, languages, etc.

Current state of Real-Time Streaming



Goal for Real-Time Streaming

Session Negotiation

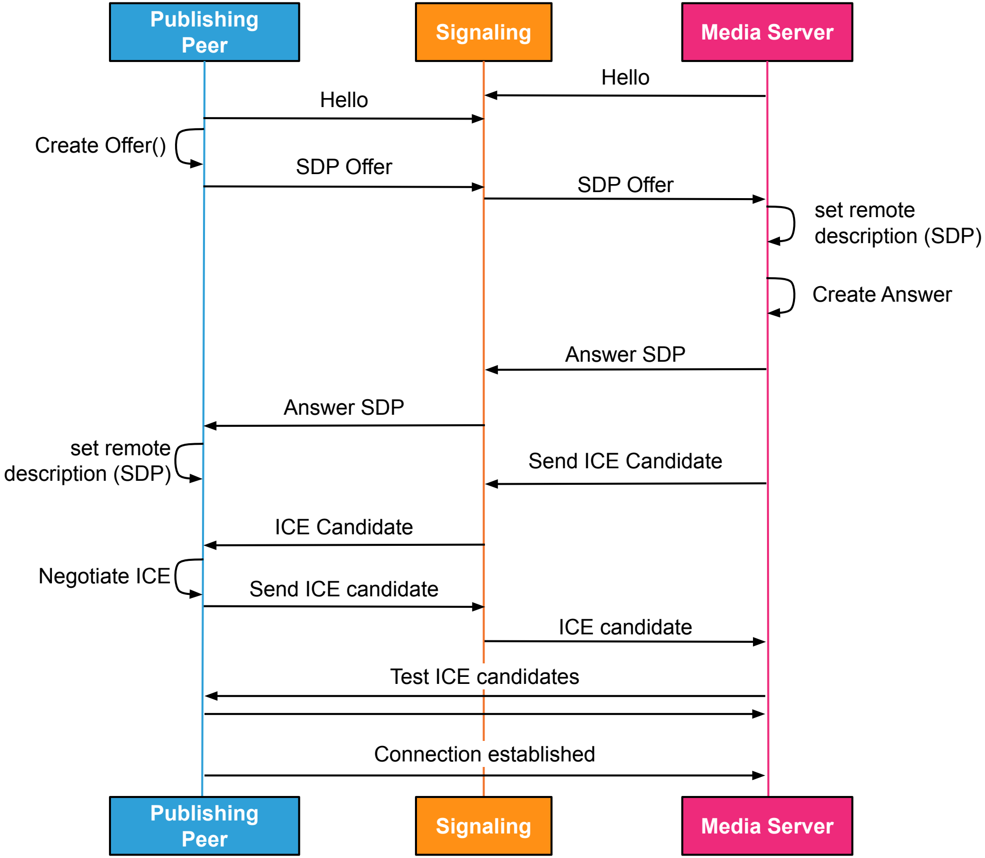
Today: The specific transport method for signaling and session negotiation is left up to each application developer

What's needed: Standard and specific transport method for signaling and session negotiation

Goals:

- Increase adoption by allowing participants to easily publish and subscribe to each other
- Minimize session establishment time while maintaining flexibility

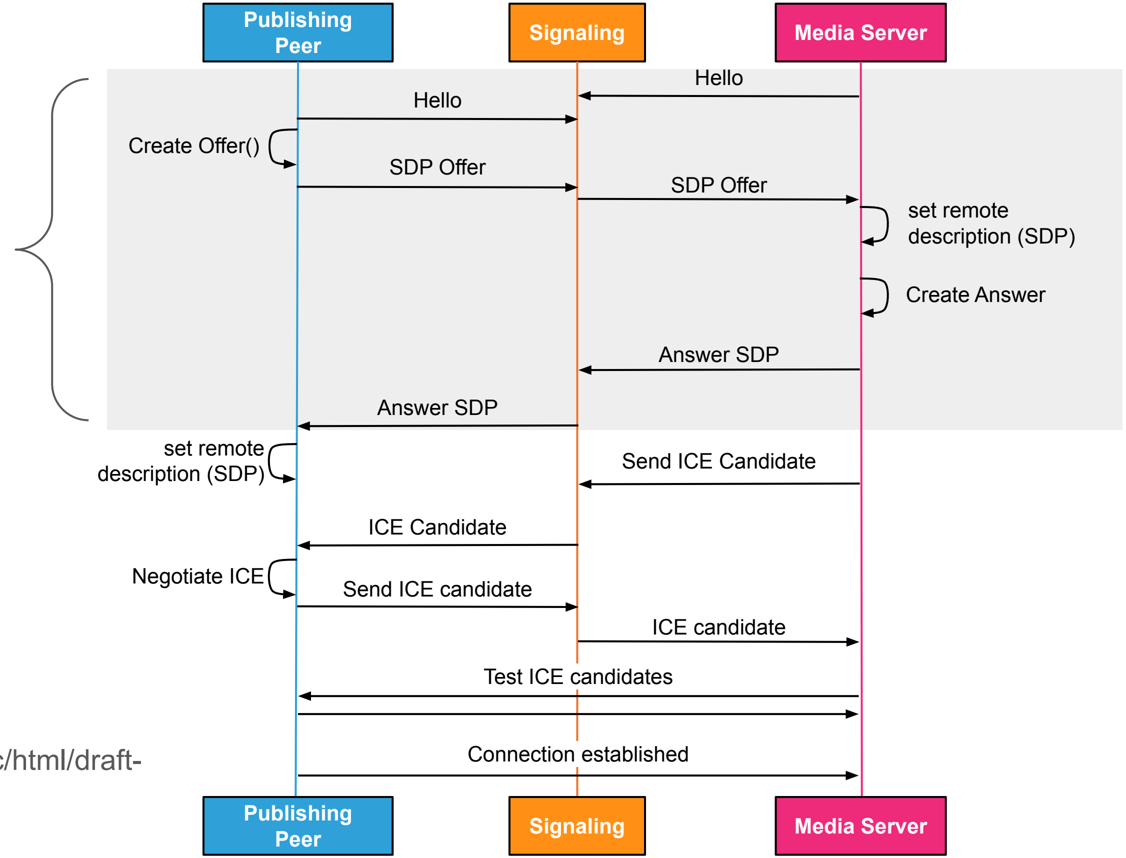
Current State - No Specific Signaling or Transport Method



WebRTC HTTP-base Ingest Protocol (WHIP)

WHIP Scope

- Publish

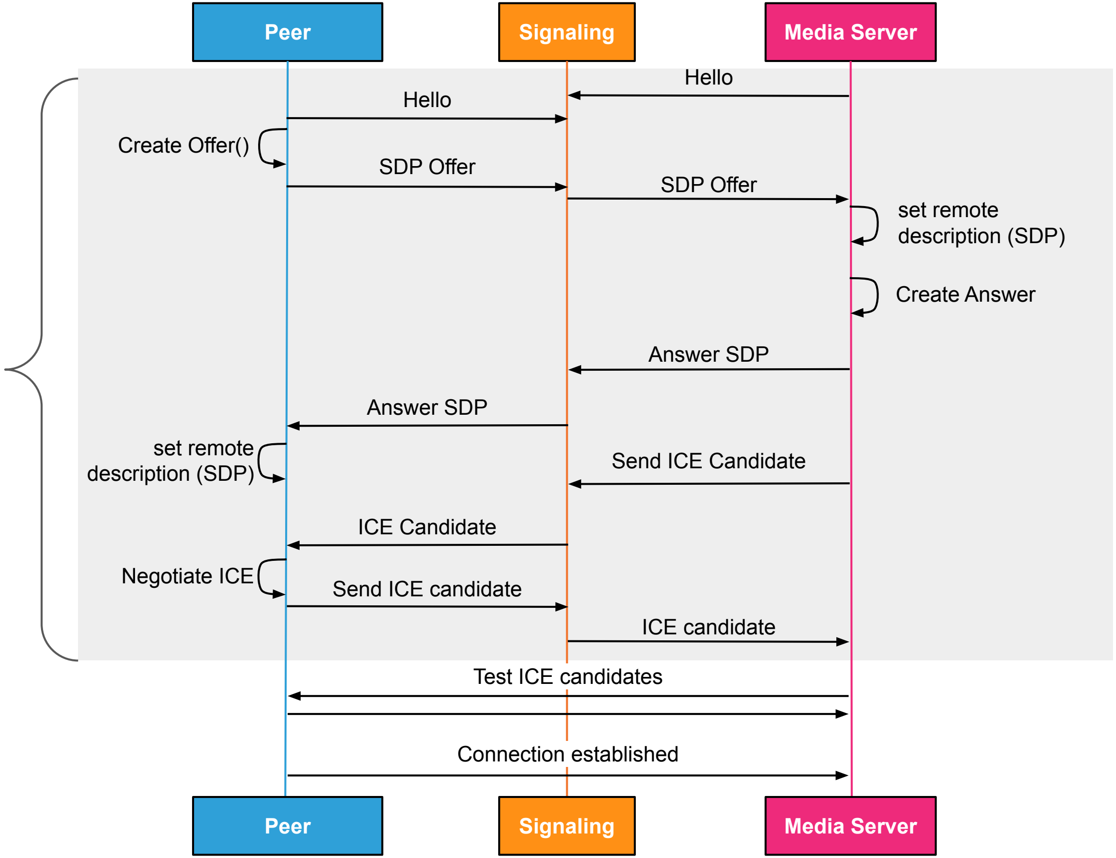


https://datatracker.ietf.org/doc/html/draft-murillo-whip-00

WebRTC HTTP Session Negotiation Protocol (WHSNP)

WHSNP Scope

- Publish
- Subscribe
- TURN
- ICE
- RTC Config



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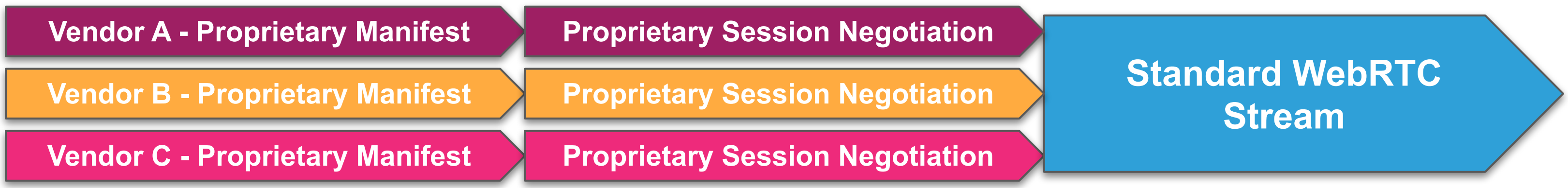


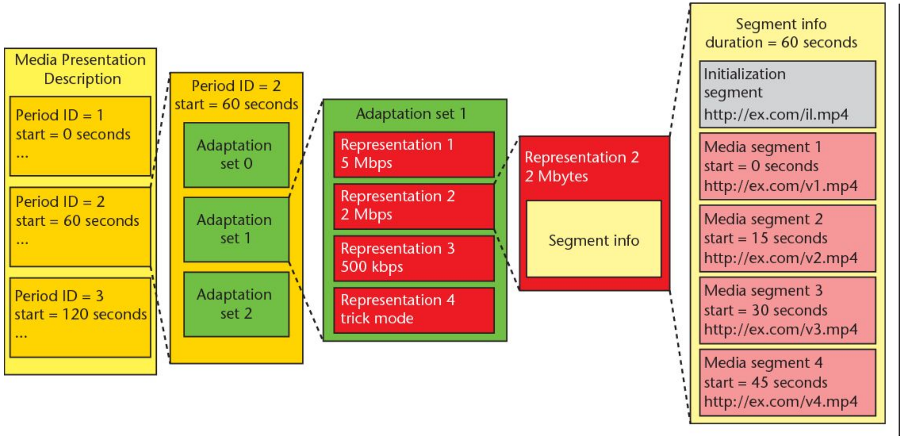
Goal for Real-Time Streaming

Can we borrow your MPD for a minute?

Why would we want to use the MPD?

- Widely adopted as an industry standard
- Solves many perceived needs
- Allows code reuse
- Adds new use cases to DASH

DASH extension for real time interactive streaming!



WebRTC Representation

- Mime type: application/webrtc
- URL for session negotiation endpoint with required content ID
- Includes audio and video
- Audio language is specified using lang= in the adaptation set
    - There is not resource duplication since audio streams can be separated from video
- Codecs are signaled in WebRTC SDP (session description protocol)
    - Scalable video codecs are starting to be supported
- Captions can use CEA-608 and languages specified in adaptation set
- Do we need multiple representations?
    - Compatibility
    - Manual bitrate selection

Events and Timed metadata

Current state:

- Delivery of ad insertion markers, captions, other timed data is not specified
- Vendors use WebRTC's reliable bi-directional data channel to send this data via proprietary messages

Goal:

- Use standard DASH in-band events and MPD events, such as SCTE-35 events, sent over the data channel
- Custom events continue to use a standard format including schemeIdUri, etc.

DASH and WebRTC use cases

- Initiate a complex WebRTC session with multiple streams
  - Using WebRTC representations with multiple adaptation sets for camera angles, etc.
- Fallback from WebRTC to HTTP streaming
  - Include Adaptations for both WebRTC and HTTP streaming
  - Players can prefer WebRTC and fallback to HTTP when WebRTC does not work
- Linear channels with interactive programs
  - Use WebRTC periods for interactive programs that require real time streaming
- Co-watching synchronized streams with audio/video chat
  - Live streams over WebRTC/HTTP or VOD over HTTP
  - Use MPD events for synchronization to fix timing issues caused by personalized SSAI
  - MPD will include information about connecting to the WebRTC-based interactive endpoint

Current WebRTC Security

Current Options:

- Secure Real-time Transport Protocol (SRTP) - Provides encryption, message authentication and integrity, and replay attack protection to the RTP data in both unicast and multicast applications

Future Options:

- Insertable Stream to E2EE - https://webrtc.github.io/samples/src/content/insertable-streams/endtoend-encryption/

DRM

Current Real-Time Options:

- WebRTC DataChannel to MSE/EME
- WebSockets to MSE/EME (**head-of-line** blocking)

Future Options:

- Insertable Stream to MSE/EME
- WebTransport to MSE/EME

MSE / EME limitations:

- CMAF isn’t ideal for real-time streaming. It’s a storage container format vs streaming format
- Current MSE/EME has some latency issues in different browsers.

DRM Cont.

Recommended Options:

- WebCodecs + EME (without CMAF packaging).
- DataChannel or WebTransport or Insertable Streams to WebCodecs and EME (without CMAF packaging).
- Client side demuxing is possible.
- AES-128 CBCS - Only Elementary stream is encrypted.
- Can follow a single encode and encryption path and multi-package approach compatible with DASH

Ad Insertion

Client-side

- SCTE event via Data Channel
- Ads via DASH or HLS and switch from WebRTC

Server-side

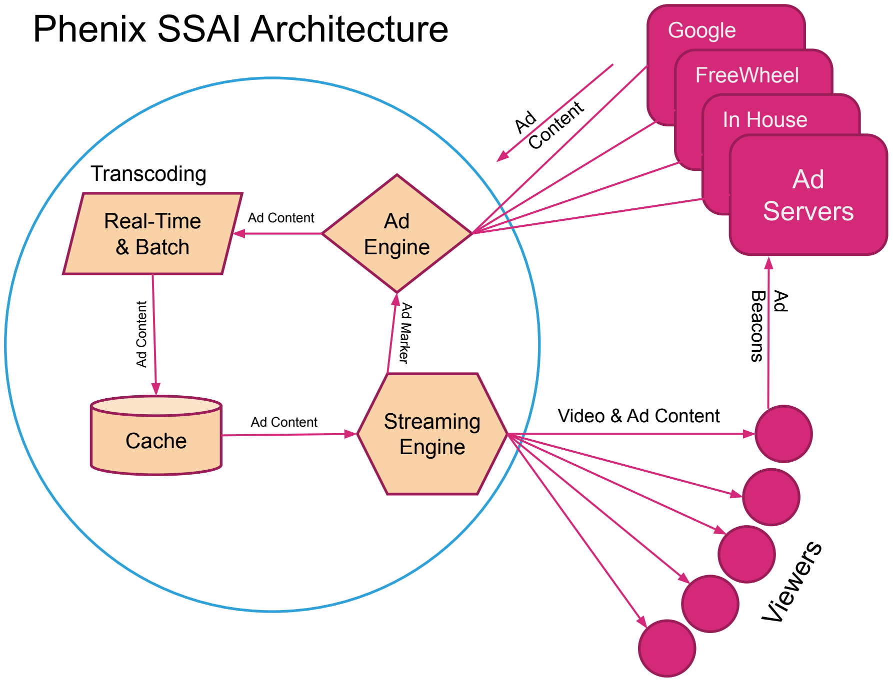
- Ad inserted directly into the WebRTC stream

Server-Side Ad Insertion in Real-Time

SSAI Situation Matrix

Ad Inventory Lead Time	Days	Seconds	None
Retrieval	Pre-fetch	Just-in-time	Real-time
Transcoding	Batch	Just-in-time	Real-time
Caching	Pre-cached	Deliver & Cache	Deliver & Cache Cache Miss => Default Ad

Phenix SSAI Architecture



Next steps

- Building industry awareness
- Identify standards development organizations for these efforts
- Industry review & feedback
  - Session Negotiation
  - DASH Interop
  - WebRTC Security

Questions?