WebRTC audio codecs for interoperability with legacy networks.

draft-marjou-rtcweb-audio-codecs-for-interop-01

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The context

draft-ietf-rtcweb-audio-01 currently states that:

3. Codec Requirements

- To ensure a baseline level of interoperability between WebRTC clients, a minimum set of required codecs are specified below. **While this section specifies the codecs that will be mandated for all WebRTC client implementations, it leaves the question of supporting additional codecs to the will of the implementer.**
- WebRTC clients are REQUIRED to implement the following audio codecs.
  - Opus [RFC6716], with any ptime value up to 120 ms
  - G.711 PCMA and PCMU with one channel, a rate of 8000 Hz and a ptime of 20 - see section 4.5.14 of [RFC3551]
  - Telephone Event - [RFC4733]

The problem

There is a real need for some additional codecs that are already deployed in billions of voice devices:

- AMR
- AMR-WB
- G.722

On one side, draft-marjou-rtcweb-audio-codecs-for-interop-01 highlights the drawbacks of transcoding:

- Cost issues
- Intrisic quality degradation
- Degraded interactivity due to increased latency
- Effiency of codec like AMR-WB over mobile radio access

On the other hand, it is recognized that implementing additional codecs in the browsers generate additional costs.

So far, draft-ietf-rtcweb-audio-01 does not encourage implementing these codecs at all, which will results in massive transcoding.

Stronger statements about these codecs are thus needed but with strong care to not put unjustified additional costs on browsers.

The proposed way-forward

The proposed way forward intends to be a compromise between browsers costs and network transcoding costs from a codec implementation perspective:

•Improve interoperability by stronger statement in draft-ietf-rtcweb-audio-01 recommending the supports of these codecs but make them mandatory only when cost impact on the browsers can be strongly limited : codecs already supported on the devices and no additional license costs.

•Add the following sub-section related to AMR, AMR-WB and G.722 in draft-ietf-rtcweb-audio-01. (There is a zoom on it in the next slide).

3. Audio Codecs

•3.1. Required Codecs

To ensure a baseline level of interoperability between WebRTC clients, a minimum set of required codecs are specified below. WebRTC clients are REQUIRED to implement the following audio codecs.

- Opus [RFC6716], with any ptime value up to 120 ms
- G.711 PCMA and PCMU with one channel, a rate of 8000 Hz and a ptime of 20 - see section 4.5.14 of [RFC3551]
- Telephone Event - [RFC4733]

•3.2. Additional Codecs

AMR-WB, AMR and G.722 codecs SHOULD be implemented by WebRTC end-points to avoid transcoding costs and quality degradations towards legacy fixed and mobile devices and allow interworking with enhanced voice quality (rather than fall back to G.711 narrow band voice).

WebRTC browsers on devices for which the implementation of AMR is mandatory for voice services MUST allow AMR to be negotiated and used at WebRTC level provided it is ensured that no additional license fees are required.

WebRTC browsers on wide-band devices for which the implementation of AMR-WB is mandatory for voice services MUST allow AMR-WB to be negotiated and used at WebRTC level provided it is ensured that no additional license fees are required.

WebRTC browsers devices for which the implementation of G.722 is mandatory for voice services MUST allow G.722 to be negotiated and used at WebRTC level.

ANNEX: Zoom on 3.2 Additional Codecs

3.2. Additional Codecs

To ensure an enhanced level of interoperability between WebRTC clients, AMR-WB, AMR and G.722 codecs SHOULD be implemented by WebRTC end-points to avoid transcoding costs and quality degradations towards legacy fixed and mobile devices and allow interworking with enhanced voice quality (rather than fall back to G.711 narrow band voice).

WebRTC browsers on devices for which the implementation of AMR is mandatory for voice services MUST allow AMR to be negotiated and used at WebRTC level provided it is ensured that no additional license fees are required.

WebRTC browsers on wide-band devices for which the implementation of AMR-WB is mandatory for voice services MUST allow AMR-WB to be negotiated and used at WebRTC level provided it is ensured that no additional license fees are required.

WebRTC browsers devices for which the implementation of G.722 is mandatory for voice services MUST allow G.722 to be negotiated and used at WebRTC level.