What WebRTC text says today:

The duration parameter indicates the duration in ms to use for each character passed in the tones parameters. The duration cannot be more than 6000 ms or less than 40 ms. The default duration is 100 ms for each tone. The interToneGap parameter indicates the gap between tones. It *MUST* be at least 30 ms. The default value is 70 ms.

Proposed text to put in rtcweb audio draft:

“WebRTC endpoints generated events MUST have duration of no more than 8000 ms and no less than 40 ms with the recommended default duration of 100 ms for each tone. The gap between events MUST be no less than 30 ms with the recommended default duration of 70 ms.

WebRTC endpoints[1], such as browsers, will not do anything with RFC 4733 tones sent from the gateway, except gracefully drop them. There is currently no API to inform JavaScript about the received DTMF or other RFC 4733 tones.”

Proposed placement:

In the WebRTC docs. That does run counter to one working group participant’s preferences:

[1] A WebRTC endpoint is either a WebRTC browser or a WebRTC non-browser. It conforms to the protocol specification. (from draft-ietf-rtcweb-overview)