Google's WebRTC Will Bring Real-Time Communications To The Browser

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*** Update, follow-up post added ***

Interesting news:
http://sites.google.com/site/webrtc/home

The WebRTc project by Google aims to enable web browsers with RTC capabilities over JavaScript APIs. For me this may just be the incentive to pick up JavaScript again. For FMS this means even more pressure outside its core function of streaming video.

Some key points and features about WebRTC:
- NAT and firewall traversal
- choice of protocols such as SIP, XMPP and others
- VP8 video
- iSAC, G.711, G.722 and iLBC audio codecs
- acoustic echo cancellation (AEC), automatic gain control (AGC), noise reduction, noise suppression and hardware access
- license allows use of other codecs
- Google Talk will move to WebRTC
- royalty free use of all included codecs

From the WebRTC blog:
"Today, we are making available WebRTC, an open technology for voice and video on the web. With WebRTC, we'd like to make the browser the home for innovation in real time communications.
With WebRTC, we are open sourcing the voice and video engine technologies from our acquisition of GIPS, giving developers access to state of the art signal processing technology, under a royalty free BSD style license. This will allow developers to create voice and video chat applications via simple HTML and JavaScript APIs."

Exiting times, and you can get involved in this very new project.
If this doesn't kick FMS into gear for stepping up its efforts for RTC based apps then nothing will. I'm not holding my breath.