



XMPP

XEP-0266: Codecs for Jingle Audio

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This document describes implementation considerations related to audio codecs for use in Jingle RTP sessions.

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1 Introduction

Jingle RTP Sessions (XEP-0167)¹ defines the Jingle (XEP-0166)² signalling exchanges needed to establish voice chat and other audio sessions using the Real-time Transport Protocol RFC 3550³; however, it does not specify which audio codecs are mandatory-to-implement, since the state of codec technologies is more fluid than the signalling interactions. This document fills that gap by providing guidance to Jingle developers regarding audio codecs.

Because codec technologies are typically subject to patents, the topics discussed here are controversial. This document attempts to steer a middle path between (1) specifying mandatory-to-implement technologies that realistically will not be implemented and deployed and (2) providing guidelines that, while realistic, do not encourage the implementation and deployment of patent-clear technologies.

2 Basic Considerations

The ideal audio codec would meet the following criteria:

Quality The encoding quality is acceptable for deployment among XMPP users.

Packetization The specification of the codec clearly defines packetization of data for sending over RTP.

Availability The codec can be implemented on a wide variety of computing platforms and is commonly used in Internet or other systems.

Patents The codec is patent-clear. The term patent-clear does not necessarily mean that no patents have ever been applied for or granted regarding a technology, or that the technology is completely free from patents (since such a judgment is nearly impossible to make, and is outside the purview of the XMPP developer community and the XMPP Standards Foundation); the term means only that those who implement the technology are generally understood to be relatively safe from the threat of patent litigation, either because any relevant patents have expired, were filed in a defensive manner, or are made available under suitable royalty-free licenses. (Although most XMPP developers would prefer to implement codecs that are patent-clear, such options are not always widely implemented and deployed.)

Unfortunately, not all codecs meet those criteria. In the remainder of this document we discuss the audio codecs that are most appropriate for implementation in Jingle RTP applications.

¹XEP-0167: Jingle RTP Sessions <<https://xmpp.org/extensions/xep-0167.html>>.

²XEP-0166: Jingle <<https://xmpp.org/extensions/xep-0166.html>>.

³RFC 3550: RTP: A Transport Protocol for Real-Time Applications <<http://tools.ietf.org/html/rfc3550>>.

3 Codecs

This section is non-normative. Future versions of this specification might provide information about additional codecs not listed here.

3.1 G.711

G.711 refers to the Pulse Code Modulation (PCM) codec defined in [International Telecommunication Union \(ITU\)](#) ⁴ recommendation G.711, which is widely used on the public switched telephone network (PSTN) and by many voice over Internet Protocol (VoIP) providers. There are two versions: the μ -law ("U-law") version is widely deployed in North America and in Japan, whereas the A-law version is widely deployed in the rest of the world. The following table summarizes the available information about G.711.

Quality	Packetization	Availability	Patents
Good quality; no wide-band mode.	See RFC 5391 RFC 5391: RTP Payload Format for ITU-T Recommendation G.711.1 < http://tools.ietf.org/html/rfc5391 >..	Commonly deployed in both PSTN and VoIP systems.	Developed in 1972; patents have expired.

3.2 Opus

The Opus codec was developed within the IETF's [Codec Working Group](#) and has been published as [RFC 6716](#) ⁵. In essence it combines the best features of CELT (developed by Jean-Marc Valin, the creator of Speex) and SILK (created by and widely used in the Skype service). The following table summarizes the available information about Opus.

⁴The International Telecommunication Union develops technical and operating standards (such as H.323) for international telecommunication services. For further information, see <<http://www.itu.int/>>.

⁵RFC 6716: Definition of the Opus Audio Codec <<http://tools.ietf.org/html/rfc6716>>.

Quality	Availability
Extremely high quality; can be used for wide-band audio; very robust in the face of packet loss.	<p>See RFC 3551 for details. Designed to be used in RTP to carry audio and video. It is a variable-bitrate, constant-quality, wide-band audio codec. It is based on the ITU-T G.729 standard. It is a good choice for applications where high quality audio is required and where the network can support the variable bit rate. It is also very robust in the face of packet loss. It is widely used in VoIP applications. It is also used in many other applications where high quality audio is required. It is a good choice for applications where high quality audio is required and where the network can support the variable bit rate. It is also very robust in the face of packet loss. It is widely used in VoIP applications. It is also used in many other applications where high quality audio is required.</p>

3.3 Speex

According to the speex.org website, the Speex codec is "an Open Source/Free Software patent-free audio compression format designed for speech". Speex was developed by Jean-Marc Valin and is maintained by the [Xiph.org Foundation](http://Xiph.org). The following table summarizes the available information about Speex.

Quality	Packetization	Availability	Patents
Good quality; optimized for voice; can be used for wide-band audio.	See RFC 5574: RTP Payload Format for the Speex Codec	Freely downloadable under a revised BSD license at http://tools.ietf.org/html/rfc5574 ..	Designed to be patent-clear.

4 Guidance for Implementers

This section is non-normative.
 Given that Opus and G.711 are patent-clear, freely implementable, and commonly deployed,

implementers are encouraged to consider including support for both codecs in audio applications of Jingle RTP sessions. Discussion on the jingle@xmpp.org mailing list indicates a slight preference for G.711 because it is easily available and so widely deployed (e.g., in SIP networks and the PSTN). Opus has effectively superseded Speex, and implementers are strongly encouraged to include support for Opus rather than Speex among the "open" codecs.

5 Mandatory-to-Implement Codecs

As of January 2013, this document makes the following recommendations:

1. Jingle clients **MUST** implement G.711 (i.e., both PCMU and PCMA) and **SHOULD** implement Opus.
2. Gateways between Jingle networks and other networks (e.g., SIP networks and the PSTN) **MUST** implement either PCMU or PCMA (and preferably both).

Naturally, clients and gateways can implement additional codecs, such as those listed in this document.

6 Security Considerations

For security considerations related to Jingle RTP sessions, refer to XEP-0167. This document introduces no new security considerations. See also the security considerations described in the relevant codec specifications.

7 IANA Considerations

This document requires no interaction with the [Internet Assigned Numbers Authority \(IANA\)](http://www.iana.org)⁶.

⁶The Internet Assigned Numbers Authority (IANA) is the central coordinator for the assignment of unique parameter values for Internet protocols, such as port numbers and URI schemes. For further information, see <http://www.iana.org/>.

8 XMPP Registrar Considerations

This document requires no interaction with the [XMPP Registrar](#)⁷.

9 Acknowledgements

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⁷The XMPP Registrar maintains a list of reserved protocol namespaces as well as registries of parameters used in the context of XMPP extension protocols approved by the XMPP Standards Foundation. For further information, see <https://xmpp.org/registrar/>.