

## 8 Bibliografía

- [1] Colin Perkins: “RTP: Audio and Video for the Internet”, Addison Wesley
- [2] William C. Hardy: “VoIP Service Quality. Measuring and Evaluating Packet – Switched Voice”, Ed. McGraw-Hill
- [3] IETF: RFC 1889: RTP: A Transport Protocol for Real-Time Applications
- [4] IETF: RFC 3550: RTP: A Transport Protocol for Real-Time Applications
- [6] ITU-T G.723.1 (03/96). Speech coders: Dual rate speech coder for multimedia communications transmitting at 5,3 and 6,3 kbits/s
- [7] H.Schulzrinne, S.Casner, R.Frederick, V.Jacobson, “RTP: A transport Protocol for Real-Time Applications”, RFC 3550, July 2003.
- [8] H.Schulzrinne, “RTP Profile for Audio and Video Conferences with minimal control”, RFC 3551, July 2003.
- [10] Kohei Fujimoto, Shingo Ata, Masayuki Murata: “Adaptive Playout Buffer Algorithm for Enhancing Perceived Quality of Streaming Applications”.
- [11] R Ramjee, J.Kurose, D.Towsley and H.Schulzrinne: “Adaptive playout mechanisms for packetized audio applications in wide area networks”
- [12] M<sup>a</sup> Guadalupe Amores García: “Evaluación de Algoritmos de Playout mediante Emulación”, Escuela Superior de Ingenieros, Sevilla.