Teamspeak Audio Conference with Telephone Backup for Synchronous Online Seminars

Andreas Bischoff, http://prt.fernuni-hagen.de/~bischoff/

Most of today’s internet users are connected to it by NAT routers. Network Address Translation (NAT) provides a local network with private non-routable IP addresses. For outgoing connections NAT translates all local to public addresses. For incoming requests NAT is often a problem. Users of audio conferencing tools behind a NAT router have to open ports for incoming connection manually. To avoid any effort for our students we have integrated the Teamspeak [1] audio conference (Voice over IP) software into our online seminar environment [2]. Teamspeak was initially developed to be a 'NAT aware' solution for audio communication in team-based multiplayer games. The Teamspeak server software is distributed for Linux and Windows for free but with a closed source license. Teamspeak clients are available for Windows, Linux and Mac OS X (TeamSpeex). Since Teamspeak is using the audio codecs CELP (5.1 Kbit/s - 6.3 Kbit/s), GSM (14.8 Kbit/s - 16.4 Kbit/s) and Speex (Speex 3.4 Kbit - 25.9 Kbit) a modem line is sufficient to use our online seminar environment.

We have connected a Teamspeak client via a virtual audio device to a SIP Softphone (SIP, Session Initiation Protocol, RFC 3261)[4]. This modified client is used as a backup to access the audio conference over the conventional telephone network. In case of a broken internet connection the users are able to use a standard or a mobile phone to reconnect to the audio conference and continue their seminar presentations.


