A Teamspeak Asterisk Gateway as Backup solution for Audio Conferencing

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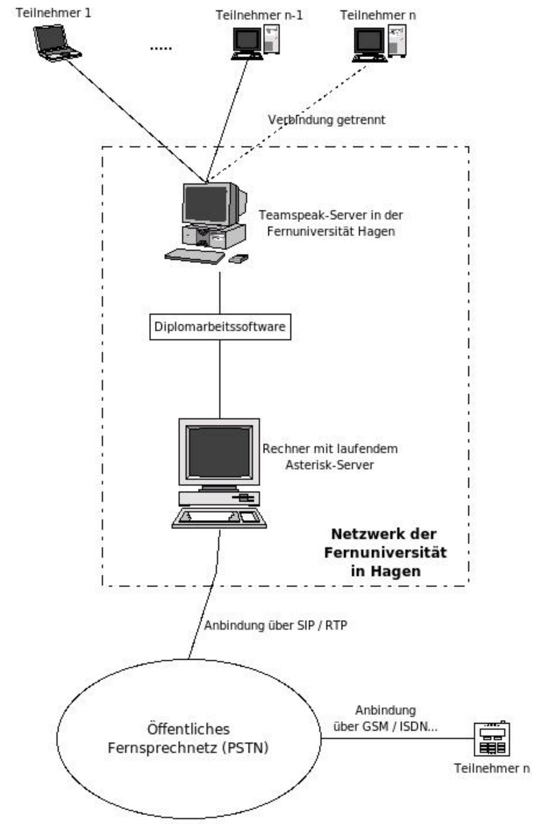

Figure 1: communication structure

To provide our students with a convenient backup solution for our Teamspeak based audio conferencing software for online seminars, we have developed a gateway software to the Internet based voice over IP (VoIP) telephony software Asterisk. Asterisk supports the so called session initiation protocol (SIP), which is the standard of Internet telephony and connects via SIP providers to the fixed line networks. Teamspeak supports the open source speex codec. An open source library (TeamBlibbityBlabbity) which supports the Teamspeak communication protocol is available. Asterisk supports the speex codec too, and is equipped with an interchange protocol (IAX2, InterAsterisk eXchange). This combination makes it possible to connect both services to realize an audio conference between Teamspeak participants and fixed line or cellphone users in realtime. Nevertheless the lack of well documented interfaces to the Teamspeak protocol requires some reverse engineering with Wireshark to figure out the Teamspeak protocol, especially separate codec payload part from the UDP protocol part of the communication. In depth knowledge of the underlying UDP packet structure and size is required to solve the task. Especially the size of a single audio UDP RTP frame differs from the size of an IAX frame. The chosen interface to Asterisk (IAX2) makes the solution independent from Asterisk versions. The Asterisk software is under heavy development, other interfaces like the AGI (Asterisk Gateway Interface) are subject to change and require a recompilation with every new version. The developed application provides a convenient way to connect both services automatically. In case of a calling student in our seminars he will be connected automatically to an ongoing Teamspeak audio conference.

[1] Christian Sixel. Realisierung der Anbindung einer SIP-basierten VoIP-Telefonie-Software an eine Teamspeak-Audiokonferenz unter Verwendung des IAX-Protokolles als automatischer Dienst unter Linux. Diploma thesis, Feb 2009

[2] http://www.asterisk.org

[3] http://www.teamspeak.com