SIP-RTSP CONVERGENCE: RTSP-C

SUMMARY

In this study, using Session Initiation Protocol (SIP) as transport and placing Real Time Streaming Protocol (RTSP) capabilities in the SIP message body, a media control model has been introduced for Voice Over IP (VOIP) networks. With the recent developments on both VOIP and IPTV technologies, convergence of these two technologies become one of the most important steps in the evolution of telecommunication. This new convergence model, RTSP-C, targets the interoperability of the two leading protocols of each technology: SIP and RTSP. The new convergnece model also resolves some open points on media control request authentication and session presentation (SDP) exchange. This new model is also valid for NAT Traversal methods applicable to SIP while it lifts the necessity of NAT Traversal methods for RTSP. The major advancement this model provides is: it makes the media control method/state information available to SIP. By doing that, this model enables the development of new streaming based SIP services. In this project content a Video on Demand (VoD) system is developpped to instantiate the new convergence model. The implementation validated the operability of RTSP-C convergence model. The comparison of the results with other models on literature showed that the model provided adequate solutions on the pre-determined problems.