EUNIS 2008: Heterogenous SIP+E-num VoIP infrastructure at Comenius University, Bratislava

1st RNDr. Martin Domany[1], 2nd Ing. Ladislav Ivancik[2]

[1] Comenius University in Bratislava, Safarikovo namestie 6, Bratislava, Slovakia, martin.domany@rec.uniba.sk
[2] Comenius University in Bratislava, Safarikovo namestie 6, Bratislava, Slovakia, ladislav.ivancik@rec.uniba.sk

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1. EXECUTIVE SUMMARY

Comenius University has built quality network infrastructure based on the dedicated optical fibers and the Ethernet 100/1000Mb/s technology. The VoIP service ranks among those using this high capacity computer network. The university VoIP system is based on the open standards and technologies, such as SIP, Enum, Sip Express Router, Asterisk. The system unites in one functional entity the standard telecommunications services based on the analogue and digital PBX and those based on the VoIP. The use of Enum allows for the possibility of making free telephone calls within the university, as well as to other destinations, e.g. in the Slovak Republic, to the Czech Republic, or worldwide. It offers the possibility of centralizing the university connection to the public telephone network and enables routing of individual calls via the cheapest possible route.

2. Background

The university VoIP system is based on the Sip Express Router application server. It serves as a central SIP registrar and proxy server. The users' IP telephones or FXS gateways are registered to it. The server provides call routing among the individual PBX within the university; calls to the public network recorded by Enum can be routed either via the VoIP or through a telecommunications operator. The PBXs of the individual university parts are connected to the central server two-way, either via the analogue/digital gateway or via the native VoIP module, depending on their type. The central SIP server is connected to the telecommunications operator by the IAX2 protocol through the Asterisk application server.

3. Conclusions

It is evident that the VoIP is becoming a dominant telecommunications services solution. When building up the completely new telecommunications infrastructure, the pure VoIP solution seems to be optimum. However, hastened migration from the classical solutions to the VoIP is not always effective and can bring about a lot of problems. The migration to the VoIP as the only solution would be expensive for the university, as a lot of its parts operate digital telephone exchanges which are relatively new and are fully efficient and reliable. Our solution enables us to preserve the current telecommunications infrastructure, as well as to integrate it with the VoIP.