VoIP in Applications for Wireless Access

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Voice over IP (or VoIP) is a common term to refer to the different protocols that are used to transport real-time voice and video and the necessary signaling by means of the Internet Protocol (IP). H.323 [1] is an ITU-T standard for real-time voice and video communication over packet networks. During the past two years it has become the standard on the Internet for VoIP. Another relatively recent example is SIP (Session Initiation Protocol) [2] for establishing multimedia sessions, that has been adopted as an RFC by the IETF. The cost to transmit digital information end-to-end is dropping dramatically, while there has been a tremendous increase in the available bandwidth. Not only has this been true in backbone networks, it has become the trend in access networks for both fixed and wireless access. The price/performance of end-user electronics is dropping while there has been a tremendous increase in computational power. As far as personal communication and mobility is concerned, we are in the position to create new applications and services [3,4] that go far beyond what telephony systems have been concerned with and able to accomplish. One of the main contributing factors is the Internet Protocol, which allows these new applications to benefit from the fact that end-user devices are now able to use multiple services over a single access. The result is that we are now able to build new interactive services, which can combine both voice and data simultaneously. While this has been widely accepted for fixed access networks, there has been quite some disagreement about what it takes to provide interactive multimedia applications over wireless access networks.

In the local area, Wireless LAN access has been available for some time and end-user devices and applications are beginning to appear in the market that address the needs of office-environments (for example the Symbol Technology "phone" which uses a wireless LAN infrastructure). In addition, there is presently a strong trend towards third-generation wireless networks for wide-area access. The basic premise is that here too, lots of bandwidth is needed to provide multimedia services and that the delivery of bandwidth should be handled by QoS-related services that are packed with the network access. Our approach is radically different by proposing to use end-to-end connectivity between hosts on top of IP (over wireless access) to bring mobile multimedia services, and that already the data-rate that is available for wireless access to Internet in today's wide-area cellular networks is sufficient to accomplish this.

We have conducted experiments with multimedia applications with integrated VoIP built on top of mobile computing devices with Internet access over both local area and wide-area cellular networks (GSM), where real-time speech was transmitted simultaneous with the exchange of Web content. The total round-trip time for speech, in the case of wireless access to the Internet with GSM data, was around 1 second. The latency that was introduced by the air link, IP network and software codecs in the mobile hosts was negligible in comparison to the transcoding by the digital modems in the base station controllers. Comparative measurements over IP networks and adjusting for additional delay in the radio link reduced the latency to levels comparable to that of mobile phones – below 200 msec. This requires a trade-off in packet-length, where short packets introduce less delay but also increased overhead in IP-headers. The perceived QoS (quality, disruptions of speech) in this case, even without header compression, ranged from good to acceptable. Compression techniques, such as "RObust Checksum-based header Compression" (ROCCO) [5] , which has been proposed in the IETF as an alternative compression algorithm to CRTP, will further improve the successful delivery of mobile multimedia applications over IP on top of wireless links in the presence of heavy packet loss. GSM speech data occupies 33 octets, to which an IP-header of 40 octets is added. ROCCO compresses this to back 2 bytes, by removing static, known and implied data in the header. Besides providing a robust link it enables us to reduce latency by further decreasing the packet size.

Our measurements concerning latency and other relevant properties indicate that it is the design of the switched network, which is responsible for bad performance (mainly latency), rather than being inherent to the solution. Furthermore, we argue that bandwidth is not the problem, as the perceived QoS was more than acceptable, even with compressed speech over a 9.6 Kbps GSM-data connection to Internet, as shown by our experiment. Bandwidth certainly helps, but the bandwidth that is available on today’s cellular networks for mobile telephony is already sufficient to bring interactive multimedia services, which allow us to blend Web content with voice. We therefore propose to reduce the protocol stack to just send IP-packets over the radio link. In the case of GSM, the radio link is RLP, where GPRS has RLC/MAC [5] . GPRS, followed by EDGE, on road towards W-CDMA, offers a packet service that will be used for providing IPv4 connectivity and thus, according to our research, allow us to deliver multimedia. This has two advantages, the first one being that no calls have to be set up prior to multimedia session, and in addition, users experience continuous connectivity. The second advantage is a simplification in the infrastructure. A router with a radio interface replaces the base station and base station controller. No transcoding has to be done in the access point, effectively removing the major part of the latency as well as significantly reducing the cost of the infrastructure.

A serious problem for developing new applications is the call model for services inherited from fixed telephony into today's cellular networks. Users do not want to accept the delays currently inherent in setting up sessions using dial-in access over wireless networks. Even if these delays in setup time are reduced, we are left with separate voice and data connections -- resulting in the same application limitations that have troubled ISDN. Unfortunately the telecom industry now proposes to extend today's call model to third-generation wireless networks. Other research proposes a new service architecture that does not suffer from these limitations. Just as ATM has proven unnecessary for the fixed network, we believe that circuit switched allocation for voice calls is unnecessary for the wireless access network. Also, we should refrain from incorporating complicated resource reservation mechanisms, since, according to our investigation, bandwidth is not the problem. Our conclusion is that we should focus on the development of packet based access points and mobile devices that are able to communicate via IP over a packet radio access network by simply running IP over the radio link protocol for the delivery of both data and multimedia services.

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