VoIP over Wireless for Mobile Multimedia Applications

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Abstract*. Voice over IP (VoIP) exploits the ability of IP to deliver multiple services over a single access link. Increased computing power and available bandwidth in end-points, reduced cost and size of electronics, as well as sensor and positioning technologies allow us to develop new interactive multimedia applications that are able to adapt to the communication context of the end-user. This is particularly important as wireless access to the Internet and connectivity between mobile artifacts can leverage these possibilities even further to bring us new ways of communication. Our work shows that, with respect to VoIP over wireless networks, bandwidth is not the problem and that QoS can match that of voice in today's cellular networks. We therefore propose to run IP directly over wireless links to bring multimedia services to mobile users. Even more importantly, this leads to a significant simplification, and consequently a cost reduction, of the wireless infrastructure.*

1. Introduction

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 alternative is SIP (Session Initiation Protocol) [2] for establishing multimedia sessions. SIP 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 of both wired and wireless links. The price/performance of end-user electronics is dropping while, at the same time, 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 link. The result is that we are now able to build new interactive services, which can combine both voice and data simultaneously. Fig. 1 illustrates the change.

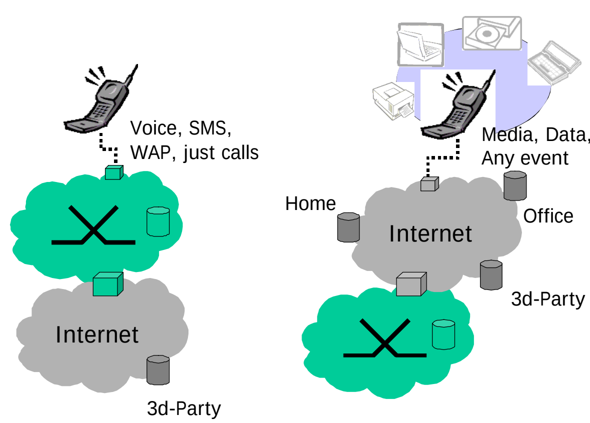

Fig. 1 Before: Telecom Up, After: Internet Up

While this change 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.

2. Research Issues

Fig. 3 illustrates the research issues that we argue must be addressed to bring mobile multimedia services over wireless networks.

- Mobile hosts, artifacts, and resources must be able to invoke *ad-hoc sessions*. JINI and Universal Plug & Play only hint at a solution.
- IP4/IPv6 Access Points that route IP-traffic between mobile devices and an IPv6 infrastructure and Mobile-IP based mobility management. In addition they must be able to route IPv4-traffic for mobiles for existing wireless packet networks, such as GPRS.
- Communication between mobile nodes must be able to carry multiple services, including real-time traffic. VoIP signifies the required functionality.

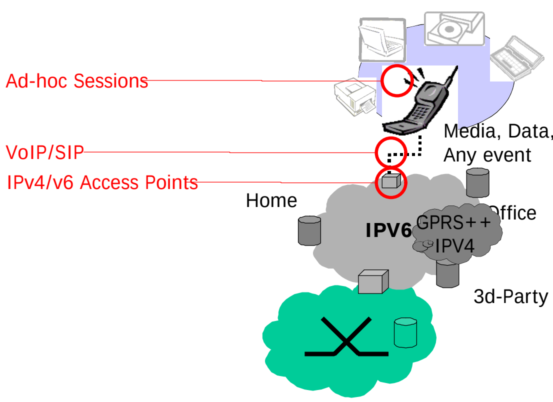

Fig. 3 Issues

This paper focuses on and proposes functionality and requirements for the delivery of real-time multimedia applications to mobile users using wireless access networks. The widely held basic premise, with respect to wireless access networks, and third generation networks in particular, is that lots of bandwidth is needed to provide these multimedia services and that the delivery of bandwidth should be handled by QoS-related services that are packaged with the network access.

Our approach is radically different because we propose using end-to-end connectivity between hosts on top of IP (over wireless networks) to bring mobile multimedia services. We believe that the data-rate already available for wireless Internet access even in today's wide-area cellular networks is sufficient to accomplish voice services.

3. Prototype

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

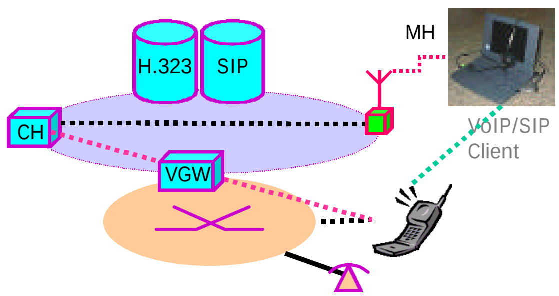

Fig. 2 Prototype

simultaneous with the exchange of Web content.

The VoIP/SIP client on the mobile host (Fig. 2) was implemented in Java (GUI) and C++ on a standard 133 MHz laptop running Windows 95, with RTP functionality from Radvision's H.323 stack. For wireless access to Internet over GSM-Data, we used an Ericsson GC-25 PC-Card-modem. A number of software CODECs were tested, among which Voxware's RT24, SC3, SC6, and Elemedia's SX1200. The following comments regarding these design choices must be made:

- **Operating System**: Windows 9X is not a real-time OS, and offers limited task priority settings. Therefore few or no other applications must be running in parallel to a VOIP-client (in our case the only other task is Internet Explorer). The TCP/IP-stack implementation in Windows introduces some extra latency in comparison to, for instance, Linux, but the effect was negligible compared to the total latency.

- **Hardware**: Full-duplex soundcards must be used. Some soundcards, such as the Soundblaster 16, emulate full duplex, thereby significantly increasing the latency.
- Speech CODECs, making up for lost packets, introduce latency, but typically not more than 40 msec worth.

4. Measurements

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, i.e. below 200 msec. This requires a trade-off in packet-length, as although short packets introduce less delay they increased overhead due to IP-headers. The perceived QoS (quality, disruptions of speech) in this case, even without header compression, ranged from good to acceptable.

5. Improvements

Compression techniques, such as "RObust Checksum-based header Compression" (ROCCO) [5], have been proposed, as an alternative compression algorithm to IP Header Compression over PPP (a.k.a. CRTP) [6]. This 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 header 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 in today's cellular networks for mobile telephony is already sufficient to bring interactive multimedia services, which allow us to blend Web content with voice.

6. Simplified Infrastructure

We therefore propose to reduce the protocol stack to just sending IP-packets over the radio link. In the case of GSM, the radio link protocol is RLP, while GPRS uses RLC/MAC [7]. GPRS, followed by EDGE, on the 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 the multimedia session, and in addition, the users experience continuous connectivity. The second advantage leads to a simplification of the infrastructure.

A router with a radio interface can replace the current base station and base station controller architecture. No transcoding has to be done in the access point, effectively removing the major contribution of the latency as well as significantly reducing the cost of the infrastructure – see also [8].

7. Extensions, Future Work

Extensions of this research regarding (1) the improvement of QoS between the mobile node and the access point and (2) providing a means to adapt the mode of communication to the current context of the user, are possible in the following areas:

1) Adapt the radio channel characteristics (W-CDMA approach). This approach calls for a mobile middleware with an API that requires the mobile to negotiate the radio channel characteristics in conjunction to setting up a session according to the Session Description Protocol [2].

2) Adapt the radio interface dynamically according to the communication context.

Research in this context is referred to as Cognitive Radio [9] and builds on Software Defined Radio [10].

3) Adapt the ongoing communication (initiated by the user user or radio channel behavior, based on mobile-IP, SIP) to the available bandwidth and radio channel characteristics. For instance the sessions could renegotiate CODECs. This is also an element of Cognitive Radio [9].

4) OS-related questions in the mobile node, such as task scheduling, TCP/IP-implementation, and handling of voice data.

5) Hardware, such as full-duplex soundcards, etc.

6) Speech CODECs, compensating for lost packets, ...

8. Conclusions

Our conclusion is that given an IP-access directly over the wireless link, mobile-IP for mobility management, and software radio, we can reuse the existing infrastructure while enhancing it further enabling the rapid introduction of new services at low cost [3,4].

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. Our research proposes a new service architecture that does not suffer from these limitations [3,4].

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 low cost 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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