RIMA: Router for Integrated Mobile Access

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ABSTRACT

Next generation wireless networks will rely heavily on packet transport for both data and voice services. In this paper, we describe the Router for Integrated Mobile Access (RIMA) system, which is the core network element of an integrated wireless packet network. RIMA acts as a Mobile Switching Center for standard cellular telephony users, and a mobile router for wireless packet data users. RIMA includes a novel call processing system that supports several types of telephony services, and a new IP mobility protocol, called HAWAII, that efficiently manages the IP mobility of data and packet voice users. In this paper, we discuss issues with providing wireless packet service, present a network architecture based on RIMA to address those issues, and describe a research prototype of the system.

I. INTRODUCTION

Packet transport service is becoming an increasingly important element in planned third generation wide-area wireless networks. As evidence, the bridge between the current second generation digital networks and third generation networks is seen to be the set of newly emerging standards for wireless packet data. However, third generation wireless networks will use packet transport for more than just traditional data applications; packet transport is envisioned as a critical element for voice and multimedia services as well.

The growth of packet data in wired networks is well documented in terms of the rate of Internet growth. The rate of packet data growth in terms of outdoor wireless networks to date has been negligible. Outdoor wireless packet data networks, such as Cellular Digital Packet Data [1] and others, have found limited, niche markets. Some of the reasons for the lack of growth have been limited wireless data devices, limited wireless network bandwidth, and the cost of wireless data services.

New wireless packet data standards, such as the General Packet Radio Service (GPRS) [2] and Enhanced Data Rage for GSM (EDGE) [3], promise over 100Kbps of shared packet bandwidth for wireless users. Third generation networks promise even higher bandwidth [4]. There have also been great improvements in handheld devices for wireless communications. These advances have led to predictions of explosive growth in the wireless data market. However, even if this predicted growth in the wireless data market occurs, the expected revenue from wireless voice services is still expected to be 25 times greater than wireless data services in 2003 [5].

One desirable scenario is that third generation wireless network infrastructure support packet voice. Packet voice, or voice over IP (VoIP), is a growing application on the Internet, and many telecommunication carriers are using, or planning to use, IP for voice transport over long-haul networks. The main advantage of such a network is that while it may support packet data service because it is packet based, it preserves the revenue of cellular voice services. However, the introduction of wireless packet voice service will likely take place in phases, and interworking with both circuit egress networks, such as the Public Switched Telephone Network (PSTN), and current circuit air interfaces, will be required.

In this paper, we present an experimental research system called the Router for Integrated Mobile Access (RIMA). RIMA is the central network element in a network designed to support packet data, packet voice, and circuit voice applications. The current prototype system supports wired packet transport for standard GSM [6] voice calls, and end-to-end packet transport for data applications. The purpose of the system is to show that an IP-based router can be enhanced to support integrated wireless access.

As part of RIMA, a new call processing system was designed and built to support call control for both circuit and packet voice service. This system is extended from previous work done on call processing for mobile networks [7]. Also, a new IP mobility protocol, called HAWAII [8], was designed and developed to provide efficient IP-level mobility capable of supporting both data and packet voice applications.

The remainder of this paper is organized as follows: in Section 2, we provide an architectural overview of emerging wireless packet networks; in Section 3, we present a RIMA-based wireless packet network; in Section 4 we discuss the various scenarios in which wireless packet systems will be used; in Section 5 we discuss the call processing and mobility management aspects of RIMA in detail. In Section 6, we conclude.

II. EMERGING WIRELESS PACKET NETWORK ARCHITECTURE

Emerging network architectures for wireless packet networks treat packet data as a parallel addition to wireless circuit networks. For example, the GPRS-based packet network operates in parallel to the GSM-based circuit network as shown in Figure 1.

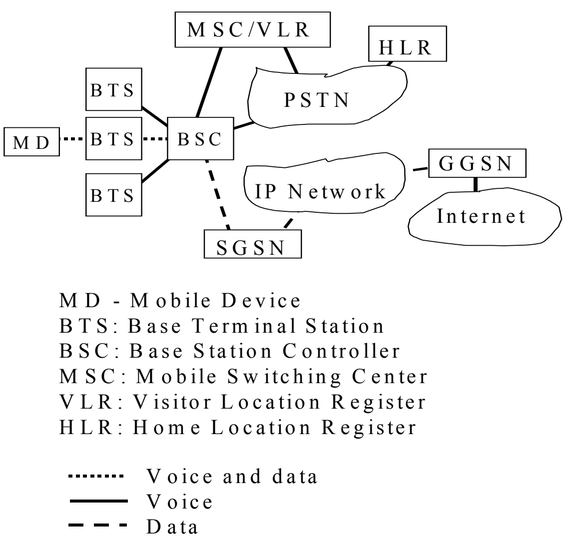

Figure 1. Simple GPRS Network

The GSM wireless circuit network consists of Base Terminal Stations (BTS) that terminate the air interface with the mobile station, Base Station Controllers (BSC) that manage the various BTS, Mobile Switching Centers (MSC) that perform call processing, handoff, and paging functions, and the Home and Visitor Location Registers (HLR, VLR) that perform mobility management and authentication functions.

The addition of packet data to the GSM network results in two additional nodes called the Serving GPRS Service Node (SGSN) and Gateway GPRS Service Node (GGSN). The SGSN performs the handoff and paging functions for packet data while the GGSN acts as a gateway router to the Internet.

In this arrangement, the packet traffic is split from circuit voice traffic at the BSC; the voice traffic is sent to the MSC, while packet data is sent to the SGSN. This leads to a complete separation of voice and data except for the VLR and HLR functionality that can be re-used. The benefits of this architecture from the viewpoint of a service provider are that the existing voice infrastructure is untouched, and packet equipment is added only as needed. However, this architecture has a high overhead in terms of cost and complexity for a service provider without legacy equipment or a service provider who just wants to provide wireless packet service.

III. RIMA-BASED WIRELESS PACKET NETWORK ARCHITECTURE

An alternative architecture based on RIMA, shown in Figure 2, is to have a single, integrated network element that supports both voice and data transport. In the case of RIMA, the network element is an IP router with additional software and hardware. Such a system must perform the functionality of the MSC, VLR, SGSN, and GGSN elements in the architecture described in Section 2. To accomplish this, a common call processing engine is needed to support call control, handoff and paging functions for circuit-access (MSC functionality), authentication and mobility management for circuit and packet-access (VLR functionality), and IP mobility support to efficiently provide packet transport, handoff, and paging (SGSN and GGSN functionality).

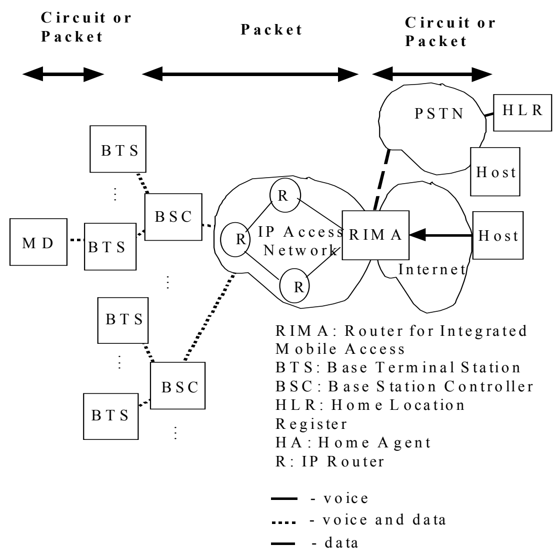

Figure 2. RIMA-based Network

This architecture has several benefits. First, a single network element is introduced that supports both voice and data applications, as opposed to multiple boxes in conventional networks. This reduces the complexity and cost of managing the network. Second, by using a packet backbone network for transport of low bit-rate voice, statistical multiplexing gains may be made reducing the cost of facilities from the BSC to the Public Switched Telephone Network (PSTN). Third, the system is well positioned to support end-to-end Internet telephony since it uses IP for voice transport. Finally, because the system is based on pure IP, it is possible to provide integrated wireless/wireline access with IP-based Quality of Service (QoS) mechanisms.

Note that the mobile access network is always packet-based as RIMA and the BSC have IP interfaces. However, as shown at the top of Figure 2, the wireless link portion could be either packet or circuit-based and the backbone network portion could be either packet or circuit-based. This leads to four ways in which a wireless service may be offered.

IV. SERVICE DEFINITIONS

The first service, termed *type 1* service, utilizes circuit access from standard cellular telephones and circuit egress to the PSTN. This is the same service provided by cellular operators today except that the access network is packet-based rather than circuit-based. In this case, since only the RIMA access network is packet-based, two media converters are required: one at the BSC and one at RIMA. The function of these converters is to modify the circuit format of the user data into packet format and establish an RTP/UDP/IP session between the appropriate BSC and RIMA. In addition, some type of voice coding function is needed in RIMA to format the voice into a compatible format with the PSTN.

The signaling and control functions implemented in RIMA must be similar to that of a standard MSC/VLR. The call processing must interact with the PSTN for connection control and database access for both Intelligent Network (IN) services and mobility management. The call processing must also interact with the base station to manage functions such as paging and circuit establishment.

In order to perform a handoff between base station controllers, the RTP/UDP/IP session between RIMA and the old BSC must be redirected to between RIMA and the new BSC. Because the access network will likely be small, this type of handoff will have similar performance as the current handoffs in circuit-switched networks.

The second service, termed *type 2* service, utilizes circuit access from standard cellular telephones and packet egress into the Internet. This is an evolution of type 1 service in which the backbone network is also packet-based resulting in cost efficiencies due to reduced tariffs, packet multiplexing, and potentially lack of voice coding. In this type of network, the BSC must act as a media gateway and convert user circuit-voice into RTP/UDP/IP packets. The call processing functionality in RIMA for type 2 service may be greatly simplified. However, performing inter-BSC handoffs is complicated.

One option, similar to handoff processing for type 1 service, is for the end host in the Internet to change the RTP/UDP/IP session to the new BSC from the old BSC. However, since the IP flow may span a long distance, this may result in a long delay causing many packets to be lost during a handoff.

Another option is for old BSC to serve as an anchor from which the session is extended to the new BSC as the user moves. While this option may result in minimal packet loss during the handoff, the efficiency of the network is reduced as the routing is no longer optimal.

A third approach is to terminate one RTP/UDP/IP session between the end host in the Internet and

RIMA and manage another session between RIMA and the BSC. This allows local IP mobility support to perform handoffs, resulting in minimal disruption. Finally, note that the end host in the Internet might actually be an Internet Telephony gateway with the actual end host being a regular phone.

The third service, termed *type 3* service, utilizes a packet air interface along with circuit-egress using the PSTN. This allows a wireless packet device to communicate with a traditional phone. In this case, the call processing in RIMA will be similar to a standard MSC/VLR. The handoffs are handled locally at the IP layer, between RIMA and the mobile device since, in this case, the mobile device terminates the IP flow. In this type of network, RIMA must act as a media gateway and perform voice coding.

Finally, we have *type 4* service where packet access is used to end-to-end. In this case, the wireless packet device has the same capabilities as any Internet data device. Call processing functions are handled in servers in the network. Mobility management is performed at the network layer.

V. RIMA SYSTEM

Our current prototype of RIMA supports type 1, 2 and 4 services: GSM voice using standard GSM handsets, base terminal stations (BTS) and BSCs, GSM service using Internet telephony for signaling to mobile phones, and packet data from an IP-end device. We only discuss type 1 and 4 services here. These services are the most likely to be deployed: type 1 services because all terminals, radio equipment, and interfaces to the PSTN do not have to be changed, and type 4 service because wireless Internet devices will be IP-based.

The high-level system architecture for RIMA is shown in Figure 3. RIMA has two main components: a call processing engine and a router core. In the case of type 1 service (circuit voice), the call processing engine in RIMA performs MSC and VLR control functions, and controls the router core that acts as a media gateway and performs voice coding. For supporting type 4 service (end-to-end packet), the router core runs standard Internet routing protocols, Mobile IP, and a new protocol called HAWAII, which extends Mobile IP to efficiently support micromobility of packet-enabled end-devices. The HAWAII daemon interacts with the call processing engine to register and authenticate users and to update location information of the IP end-devices. In the case of type 3 services, hooks are in place to allow the call processing engine to support packet voice applications.

Figure 3 illustrates the overall structure of a network based on the current implementation of RIMA. The BSC has an IP interface and translates voice and signaling information from a circuit format to packet. For signaling information the standard GSM interfaces from the BSC are tunneled in IP packets to RIMA. For voice transport, voice samples are put into an RTP/UDP/IP stream to RIMA.

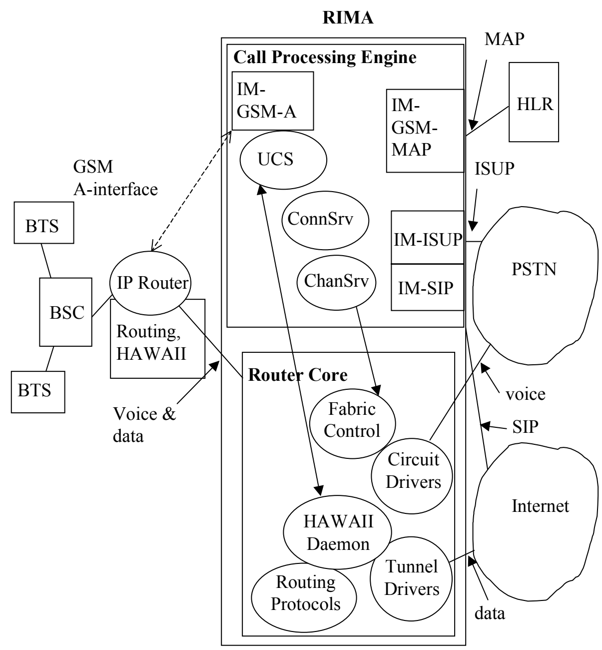

Figure 3. RIMA System Architecture

The RIMA access network may connect to the PSTN or Internet for voice communication. To connect to the PSTN, RIMA performs media transformation between RTP voice packets and PCM voice samples. Vocoding is also executed to translate between compressed wireless code (e.g. GSM speech code) and PSTN PCM code (e.g. A-law or u-law). For IP telephony, RIMA may or may not perform voice coding functions depending on whether different coding schemes are employed in the two networks.

The system currently has standard interfaces to GSM BSCs (GSM A-interface), HLRs (Mobile Application Part (MAP)) [9], and the PSTN (ISUP) [10]. For data, the system provides an IP interface to the Internet.

Details of the system operation for voice and packet data services are presented in the subsequent sections. These sections focus on the current RIMA prototype.

A. Call Processing

RIMA call processing can accommodate two types of wireless voice terminals: circuit voice wireless terminals such as GSM phones and packet voice terminals with a packet radio interface like GPRS. Our current prototype supports the former terminals, while it can be easily extended to support the latter with the help of the RIMA packet data infrastructure discussed in the next section. This section concentrates on circuit wireless voice service.

The call-processing engine is deployed on a set of single board computers to realize MSC and VLR functions. It can be also viewed as a signaling gateway from an IP telephony network. The engine consists of a collection of functionally distributed servers as shown in Figure 3. The call processing and mobility management tasks are accomplished by their collaboration. Two classes of servers are introduced: *core servers* and *interworking managers* (IM). Core servers perform call processing and mobility management tasks common to any wireless standard. Interworking managers, on the other hand, act as protocol gateways to internal core servers, isolating them from external signaling protocols thereby allowing the core servers to evolve independently of these protocols. IMs also allow core servers to accommodate different sets of standard interfaces (GSM as well as CDMA, for example).

In our system, we have interworking managers supporting the GSM-A interface to the base station controller (IM-GSM-A), GSM-MAP to the HLRs (IM-GSM-MAP), ISUP to the PSTN (IM-ISUP), and SIP to the Internet (IM-SIP).

There are three core servers: A channel server (ChanSrv), a connection server (ConnSrv), and a user call server (UCS). *ChanSrv* manages switching device resources, such as transport channels and DSPs for vocoding, allocated during call setup and deallocated during call release. ChanSrv sends media gateway control messages to instruct the BSCs and the RIMA core router on resource allocation.

ConnSrv coordinates the allocation of channel resources to establish a connection to the BSC of the cell in which the MS is currently roaming. The ConnSrv instructs the appropriate ChanSrv to reserve needed MSC channel resources and sends messages to external components via the IMs to reserve channel resources external to the MSC. For example, ConnSrv may reserve network trunk resources using ISUP control messages through the IM-ISUP.

UCS maintains information on the registration status of mobile devices currently roaming within the service area of the RIMA system and records call activities involving a particular mobile device. UCS also handles other mobility management tasks such as paging, handover, mobile user authentication, and ciphering. Some mobility management procedures implemented in UCS are reused for the IP packet data system. They include power up/down registration to the HLR, terminal authentication, and ciphering key management.

Inter-BSC handoff occurs when a client moves between serving areas under BSCs. During a handoff ChanSrv assigns new resources on a new BSC. Standard handover protocol messages are utilized to reserve radio and terrestrial resources between the BSC and the mobile phone. Additionally ChanSrv updates the BSC address on RIMA to redirect traffic to the proper destination.

The call processing system developed for wireless circuit access networks can be easily extended for wireless IP packet terminals with IP telephony clients such as SIP or H.323. This requires changes to the IM interfacing with the BSC, generically called the IM-BSC (this is the IM-GSM-A in our configuration). Transport mobility in these cases will be handled at the IP layer, so this function is no longer necessary in the IM-BSC. However, the IM-BSC will continue to perform non-transport mobility functions such as registration and authentication. In addition, the IM-BSC will terminate the Internet telephony signaling protocol used by the mobile device.

In [7], it was shown that a call processing system based on the same design principles as those that are the basis for the system presented here, can achieve high call throughput at low latencies. Both the circuit access and packet access systems will receive the same performance in terms of call processing. For the pure packet system, the call processing performance will be similar to any Internet-telephony system.

B. IP Mobility

The protocols and control algorithms for supporting packet data were designed to provide performance suitable for packet voice service in addition to traditional data applications (type 3 and type 4 services). The key component in this system is an IP micromobility protocol called HAWAII. HAWAII is integrated with the call processing portion of RIMA to allow wide area access in terms of registration and authentication of IP end devices.

HAWAII extends Mobile IP [11] to more efficiently support mobility at the IP layer. Whereas Mobile IP provides strong support for the roaming of IP end devices, the registration and handoff processing of Mobile IP is centralized and may not be fast enough to support real-time applications. In HAWAII, handoff processing is distributed and localized, resulting in fast handoffs and high scalability.

Figure 3 illustrates a simple instance of a network based on HAWAII. RIMA acts as a gateway router to the Internet and is called a domain router. Within a domain is a network of routers providing connectivity from the domain router to the BSC. Each domain has its own IP address space that is publicly routable. When a device activates itself on the network, it is authenticated and assigned an IP address from the domain DHCP server. Before assigning an address, the DHCP server checks the VLR resident in the call processing engine to verify that the client has been authenticated.

Once the client is assigned an IP address it registers with the domain router. As the registration message from the client propagates to the domain router, all intermediate routers receiving the message create a host-based routing entry for the client. When registration is complete, a route exists between the domain router and the client. Because the client uses this IP address when communicating with any desired servers, packets will be routed directly from any corresponding servers to the client through the domain router. This has two main benefits: first, there is no tunneling, and second, there is optimal routing. The fact that there is no tunneling makes it straightforward to support per flow QoS in this system using reservation protocols like RSVP.

When a client moves between IP points of attachment within a domain, in this case between BSCs, it must update its location using HAWAII updates. These updates are compatible with Mobile IP. As with the power-up registration, as the update messages propagate towards the domain router, host based routing entries are created in the intermediate routers. Once a router with a previous entry for the client receives the update message, the host based routing entry is modified and the message is no longer propagated. In this way, only the routers that must have modified entries are notified when a client has moved. This has two affects: first, handoff processing is localized and fast, and second, there is no central bottleneck in the system. Because the client maintains its IP address, and a large portion of the IP flow is unaffected by the movement, only a partial QoS re-reservation is required in order to maintain QoS after movement.

When a client moves between domains, a combination of basic Mobile IP and HAWAII procedures are used. The Mobile IP procedures are used to maintain connectivity with any corresponding hosts by creating a tunnel to the home address of the client that was acquired at initialization. The HAWAII procedures are used so that HAWAII mobility support may be used for executing fast handoffs due to subsequent movement within the new domain.

RIMA also performs paging at the IP-layer so that end hosts can be located for delivery of incoming data packets. For more details on the HAWAII protocol, please refer to [8].

The main performance concern with any wireless voice system is the voice quality. In a packet voice system, this is affected by the delay and jitter experienced by the packet voice, and the losses that occur during handoff. The jitter is eliminated through the use of playout buffers at the point where the packet stream is converted to circuit. For the networks with wireless packet access, the playout buffer is in the mobile device. For networks with wireless circuit access, the playout buffer is in the BSC. In any case, the handoffs must occur in a shorter time than the playoff buffer in order for there to be no disruption in voice communication. For this reason, the performance of the HAWAII handoffs is critical.

In [8], it was shown through extensive simulation, that HAWAII handoffs occur fast enough to support real-time packet services, such as voice or multimedia. With a playout delay of only 75 milliseconds at the mobile device, which is deemed acceptable for voice quality, a system based on HAWAII lost on average 0.3 packets per handoff.

VI. CONCLUSIONS

Future wireless networks will be based on packet switching technology. To successfully support widely used data applications on a large scale, and to allow the rate of growth seen on the wired Internet, these networks must be inherently packet based as opposed to simple packet overlay networks. Because the main source of revenue for any wireless network for the foreseeable future is voice service, the support of packet voice, or ability to interface to circuit voice air interfaces and backbone networks, is critical.

In this paper we propose a key network element for a wireless packet network to support both data and voice services. The system, called RIMA, is based on an IP router. It provides MSC functionality for voice users and IP mobility for packet data users. The voice system uses a novel, modular call processing system that allows it to flexibly control voice calls for both circuit and packet end devices. Packet mobility is provided by a new protocol, called HAWAII that greatly extends Mobile IP to provide fast handoffs at the network layer, suitable for voice service.

In the paper we describe the basic implementation of RIMA. This implementation supports voice service from standard GSM terminals, using an IP network for the wired portion of the mobile access network. It also supports end-to-end IP networking for wireless data applications. The call processing and HAWAII protocols are integrated to allow easy extension to support voice from a packet terminal.

Although the system presented here is prototyped for a network using a pure IP access network, its key concepts, namely modularizing call processing to support multiple network types, and managing mobility at the network layer in a localized manner, can be applied to other types of networks, such as GPRS.

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