Hybrid Multiplexing of Voice and Data over IP on GSM/GPRS Cellular

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Abstract— This paper investigates the benefits of using voice over IP on GSM/GPRS cellular. The work is motivated by first the complexity of having both circuit switched and packet switched connectivity on GSM/GPRS cellular and secondly by the fact that an exclusively packet based access on GSM cellular has the potential to handle more voice calls and data traffic by deploying suitably chosen hybrid multiplexing schemes that exploit the characteristics of both voice and data traffic.

Index Terms—voice, VoIP, GPRS, GSM.

I. INTRODUCTION

This paper reports work in progress on the statistical multiplexing gains obtained by transporting voice over IP over GPRS as opposed to transporting voice in GSM circuit switched channels.

Current second generation mobile systems such as GSM do not provide an effective means of offering high data rate multimedia services, with current data services being provided by a circuit switched connection. GSM networks are being upgraded to provide packet mode connection using General Packet Radio Service (GPRS) which provides data rates of up to 117.2Kbps [1].

GPRS is a packet switching service that was developed to facilitate access to IP based services by dynamically allocating radio bandwidth to users [1], [2]. Packet switching allows efficient sharing of network resources among users to reduce costs and increase the flexibility of services through statistical multiplexing.

GPRS is intended to support bursty data transactions such as Internet and WAP services [3], using GSM equipment, but is poorly suited to constant bit rate services such as video streaming [2]. Thus GPRS is better suited to frequent, small data packets up to 500 bytes and irregular packets up to a few kilobytes [2].

GPRS provides a migration path to the third generation network, providing a packet data mechanism using the existing equipment in the GSM architecture, while adding new elements interfacing to an IP based network [3], [4].

We propose that the utilisation of the air interface can be improved, if voice and data are transmitted using packet switching. The potential increase in usage can provide the cellular operator with benefits such as: increased number of simultaneous conversations, or the same number of cellular conversations while providing a greater data bandwidth. The introduction of a packet only infrastructure can provide opportunities for equipment reuse within the GPRS architecture for applications such as fixed wireless.

We will introduce the concepts of General Packet Radio Service and then discuss problems associated with transmitting voice over GPRS in general. With the sharing of radio resources between voice and data services in the GSM/GPRS system, the air interface is identified to be the performance bottleneck. We propose a strategy with regard to the gains achievable by multiplexing of voice and data packets over the GSM/GPRS air interface.

II. BACKGROUND TO GPRS

This section discusses the GSM architecture and how the transition to GPRS is achieved.

The main intention of integrating GPRS into GSM is to increase the number of connections per bearer by improved usage of the available physical channels [2]. Thus if voice is transmitted over data channels, the utilisation of the channels can be improved further.

GPRS has been standardised by the ETSI as part of the GSM phase 2+ development. GPRS usually shares the channel with GSM circuit-switched voice with one or two channels dedicated to GPRS and others being used when free channels are available [5].

Several codecs which can be used for Voice over IP (VoIP) exist, some of which support silence suppression such as G.723 and G.729 [6]. These codecs produce packets of thirty to a hundred bytes depending on the codec. With header compression and the fact that silence occurs for roughly 60 percent of the conversation the statistical multiplexing gains of the GPRS channel can be high, [7].

GPRS uses an IP backbone network with standard packet network nodes such as routers, DNS servers and firewalls, and introduced two new elements to the GSM infrastructure: the Serving GPRS Support Node (SGSN) and the Gateway GPRS Support Node (GGSN) [2], [3], [5].

The architecture of a GSM/GPRS network is shown in figure 1. Connection to the integrated services digital network (ISDN) and public switched telephone network (PSTN) is through the Gateway Mobile-services Switching Centre (GMSC).
GSM / GPRS ARCHITECTURE SHOWING CONNECTION TO PSTN/INTERNET/ISDN AND OTHER GPRS NETWORKS

Fig. 1

The GGSN serves as the interface between the GPRS core and the external public data network or other mobile networks. The GGSN is responsible for routing of mobile subscribers via the SGSN. Accounting of resources usage by user flows and mobility management is also handled by the GGSN.

The SGSN handles the terminal mobility and authentication functions and represents the GPRS switching centre by analogy to the GSM MSC. The SGSN maps packet data addresses to the International Mobile Subscriber Identity and is responsible for routing within the IP network. Resource management as well as encryption is handled by the SGSN.

Border gateways are used to interconnect GPRS IP backbones and firewalls prevent unauthorised access.

GPRS employs Gaussian Minimum Shift Keying modulation with four encoding schemes, CS-1 to CS-4. Coding Scheme 1 offers the highest level of data protection. CS-1, CS-2, and CS-3 use convolutional codes and block-check sequences of varying complexity [4]. CS-4 offers no error protection of user content and provides only error detection functionality [4], [8].

See table I for encoding schemes and their associated data rates.

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Code Rate</th>
<th>Radio Block</th>
<th>Coded Bits</th>
<th>Punctured Bits</th>
<th>Data Rate (kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>CS-1</td>
<td>1/2</td>
<td>181</td>
<td>456</td>
<td>0</td>
<td>9.05</td>
</tr>
<tr>
<td>CS-2</td>
<td>2/3</td>
<td>268</td>
<td>588</td>
<td>132</td>
<td>13.4</td>
</tr>
<tr>
<td>CS-3</td>
<td>3/4</td>
<td>312</td>
<td>676</td>
<td>220</td>
<td>15.6</td>
</tr>
<tr>
<td>CS-4</td>
<td>1</td>
<td>428</td>
<td>456</td>
<td>0</td>
<td>21.4</td>
</tr>
</tbody>
</table>

*Excludes USF and BCS bits

The radio block presented is without the multiplexing control Uplink State Flag (USF) header and error detection Block Check Sequence (BCS) trailer. The coded bits are the size of the frame after convolutional encoding. Puncturing is used to shorten the convolutional code in coding scheme 2 and 3.

One TDMA frame, as shown in figure 2, represents eight physical channels to be shared by the GSM and GPRS. Sharing strategies are implemented by the network operator or equipment vendor to allocate the GPRS resources according to set policies. The physical channels used in GPRS for packet data traffic are known as Packet Data Channels (PDCs). Logical PDCs are mapped onto physical channels using a cyclically recurring multiframe structure [2]. Six radio blocks, each containing 8 PDCs, are mapped into one 26-multiframe, and two 26-multiframes are assembled onto one GPRS 52-multiframe. Thus a 52-multiframe represents one physical GPRS channel consisting of twelve radio blocks and one idle block, each block comprising four radio bursts distributed on timeslots with the same timeslot number in consecutive TDMA frames, as shown in figure 3.

### III. CHALLENGES OF VoIP OVER GPRS

Transmitting voice over IP by using GPRS introduces many challenges that need to be carefully considered. Issues such as header efficiency and quality of the air interface affect the successful end-to-end transmission of voice.

An adequate Quality of Service (QoS) is required to transmit voice over wireless links, the main requirements being delay and bit error rate [9]. GPRS QoS profiles are currently limited to only one QoS profile per Packet Data Protocol address, with GPRS phase one not allowing QoS re-negotiation to be initiated by the MS or GGSN [3], [10]. GPRS was designed for best effort traffic and as such the specification of parameters such as delay constraints are often un-implementable on an end-to-end basis.

Other issues such as optimisation for real time handovers and fixed bandwidth reservations are also not well supported.
by GPRS. Thus a revision of the access control of the GPRS can improve the partitioning strategies by introducing variable bandwidth requirements between voice and data communication. Partitioning strategies such as complete sharing and bit rate sharing provide a high link efficiency, with both strategies being able to use all available cell capacity [1].

The two competing voice over IP control protocols – the ITU-T H.323 and the IETF Session Initiation Protocol (SIP) – both use the Real-time Transport Protocol (RTP) to transport the voice codec stream [11]. The voice packets generated by the speech coder are encapsulated in RTP, UDP and IP before being passed to the GPRS network. A speech frame has a RTP / UDP / IP header size of 40 bytes and a payload of 15–30 bytes depending on the voice codec [12]. If IPv6 is used the header size is increased to 60 bytes [12]. Due to the high overhead, header compression is required with schemes such as compressed RTP and the favoured Robust Checksum based header Compression [12].

The GPRS Medium Access Control Layer is responsible for providing efficient multiplexing of data and control signalling on the uplink and downlink. MAC procedures include the provision of Temporary Block Flow (TBF) which allow point-to-point transfer of data in one cell between the base station subsystem (BSS) and the mobile station (MS) [2]. The BSS consists of both the base transceiver station (BTS) and the base station controller (BSC). TBF releases and re-establishments during the transmission of voice packets impact on the delay of voice packets, and can be adjusted to meet end-to-end delay requirements [6]. Optimising the TBF duration is required to efficiently utilise the medium.

IV. END-TO-END SOLUTION

This section illustrates why the air interface is the bottleneck. Each base station subsystem (BSS) is virtually connected to a SGSN by means of a connection-less link on the Gb interface. The BSS does not perform flow control in the uplink direction, that is from the BSS to the SGSN [2]. Thus the SGSN buffer and link capacity is expected to be dimensioned such that uplink data is not discarded.

All GPRS Support nodes (GSNs) are connected to the IP backbone. Encapsulation is used to transfer messages between GSNs using the GPRS Tunneling Protocol (GTP) [10]. Information is transferred between the BSS and SGSN using the Base Station Subsystem GPRS Protocol (BSSGP) as shown in figure 4. IP routing mechanisms are used to transfer the packets to appropriate GSNs.

Since any transport mechanism may be used below IP the QoS and bandwidth constraints are assumed to be in the Um and Gb interfaces - illustrated in figures 1 and 4.

V. PROPOSED WORK

In order to evaluate the statistical multiplexing gains obtained by transporting voice over IP using GPRS, research will concentrate on the following aspects:

- The bottleneck of the system is seen as the Um and Gb interfaces, thus the focus of the research is on the air interface and SGSN.
- Control signalling such as that used by H.323 or SIP before and after calls will not be considered, instead focus will lie on the voice coded RTP data.

Different call scenarios will be considered such as from the MS to another MS, MS to PSTN, MS to Internet phone.

Initial modeling will consist of a basic model such as that shown in figure 5, with all sources being queued.
The voice traffic will have an average arrival rate of $a_1$ and holding time of $m_1$ while data will have an average arrival rate of $a_2$ and holding time of $m_2$. Voice data will be considered time delay sensitive and packet loss insensitive while data would be considered packet loss sensitive and time delay insensitive. An appropriate queueing model for voice and data using GPRS will be developed.

Traffic simulation over an area will be initiated according to a uniformly distributed Poisson process, a mean holding time of 120s [11] and mean talk-spurt length of 1.2s [13] will be used for the voice. A possible voice model that can be used is the 3 state Markov model which ignores double talk but models silence better [13].

Results such as the required bit rate per conversation as well as the maximum number of conversations can be deduced. The optimal delay and packet size can be calculated. The capacity at a constant level of quality can be identified in a manner similar to [11].

The encoding scheme can be varied to determine the most suitable level of protection for data, voice, and both data and voice.

The effect of various voice codecs can be simulated with a data rate particular to that codec to determine what basic throughput would be generally required.

VI. CONCLUSION

This paper has discussed our work in-progress on statistical multiplexing gains possible by transmitting both voice and data in GPRS packet data channels. The fact that GPRS is suited to frequent small data packets and that voice packets are of the order of a hundred bytes suggest that GPRS will be well suited to carrying voice over IP. Considering as much as 60% of a conversation is silence, the multiplexing gains would provide an operator with increased cell capacity, providing a better utilisation of the limited radio resources on the GSM radio interface. We have highlighted issues to consider in the transport of voice over IP using GPRS such as encoding schemes and header compression, and a method of research was proposed to simulate the multiplexing gains.

REFERENCES


