FLEXIBLE LAYER ONE FOR THE GSM/EDGE RADIO ACCESS NETWORK (GERAN)

TOMMI JOKELA
Nokia Technology Platforms, Itämerenkatu 11-13
Helsinki, FIN-00180, Finland

SHKUMBIN HAMITI
Nokia Technology Platforms, Itämerenkatu 11-13
Helsinki, FIN-00180, Finland

BENOIST SÉBIRE
Nokia Research Center, Hepingli Dongjie 11
Beijing, 100013, China

GUILLAUME SÉBIRE
Nokia Technology Platforms, Itämerenkatu 11-13
Helsinki, FIN-00180, Finland

In the Release 6 of 3GPP, a new type of physical layer has been standardised for the GSM/EDGE Radio Access Network (GERAN): the Flexible Layer One (FLO). Rather than having fixed coding schemes in specifications and corresponding implementations, FLO provides a framework that allows the layer one to be configured and optimised at call set-up. As a result, the introduction of new services can be handled smoothly without having to specify new coding schemes in each release. Together with Iu alignment, FLO enables seamless provision of the *same* services over GERAN as over UTRAN. This paper describes why FLO is needed, its architecture, and presents its performance for a Voice Over IP (VoIP) service.

INTRODUCTION

For an optimised support of new real-time services, new coding schemes have been specified in GSM/EDGE for more than a decade. While this approach yields an optimised link level performance, it is both slow (specification) and costly (implementation). For instance, the introduction of AMR-WB in Release 5 required the introduction of 38 new pages of coding schemes in 3GPP TS 45.003 alone! Hence, for a quick, optimised and cost-effective introduction of new real-time services in GERAN, it was decided to follow the same approach as in UTRAN: a flexible layer one [1] [2] [3] [4].

Through several enhancements, the radio bearers offered by FLO fulfil the IP Multimedia Subsystem (IMS) requirements in terms of flexibility and performance, but also significantly improve the link level performance of real-time IMS services compared to

GERAN Release 5. In addition to the IMS services, FLO improves the flexibility and performance of the current and future circuit-switched services.

This paper describes the architecture of FLO and shows its performance for a typical real-time packet-switched service – Voice Over IP (VoIP). The main motivations for having FLO are given in Section II, while the architecture of FLO is explained in Section III. In Section IV, the real-time performance of FLO is compared with the real-time performance of EGPRS by using an AMR-based VoIP service as an example. Finally, conclusions are drawn in Section V.

MOTIVATION

Mobile voice traffic continues to grow and already today many networks are stressed with capacity and quality challenges. Although growth in revenue will mainly come from data services, it is foreseen that voice will still generate 60% of the global operator revenue in 2006. Improvements to optimise the capacity and quality of speech services are therefore very important not only for the speech but also for the data services and that is the reason why they have been continuously introduced over the GSM/EDGE system for more than a decade: EFR, AMR-NB, AMR-WB, and AMR-NB on 8-PSK channels.

Today it is difficult/questionable to claim that no new services will be introduced on the GSM/EDGE system. With constantly increasing processing power and battery life, high quality codecs (e.g. AMR-WB+) and good quality low bit rate codecs are very likely to be introduced in the near future.

In addition, IP based real-time services, especially VoIP is appearing as an attractive service in the fixed internet. Current EGPRS coding schemes provide sub-optimal performance for real-time IP services in terms of spectral efficiency. Although the spectral efficiency may not be important in the early phase of deployment of VoIP services, it becomes very important issue when customer base expands, therefore solution to improve the efficiency are needed.

The traditional way of solving these problems has been to introduce a specific channel coding for a particular service or set of services. This has been done for all speech codecs in GSM (FR, HR, EFR, AMR-NB, etc.). The whole process (standardisation, implementation, testing) is time consuming and lead to delays in the deployment of new services.

So the key question is: how can we provide a generic solution that would improve the spectral efficiency and quality, allow for fast introduction of new services and at the same time keep the complexity manageable?

One answer to this question is FLO. With the Flexible Layer One concept we can guarantee a fast, economical and efficient introduction of new services in the long term, while keeping the complexity on a reasonable level.

ARCHITECTURE

General Principles

Instead of having in specifications a fixed set of coding schemes that are tailored and optimised for a limited number of services, FLO provides a framework that allows the coding scheme to be configured and optimised at call set up according to the QoS requirements of the service to be supported.

Protocol Architecture

In order to accommodate FLO concepts and principles, changes are required to GERAN radio protocol architecture. In GERAN *Iu mode*, the protocol architecture when relating only to FLO is depicted in Figure 1 below, where most impacts are located at and between the MAC sublayer of layer 2 and the physical layer. It should be noted that FLO is only available on dedicated basic physical subchannels (DBPSCH), and allows for data transfer in transparent, or non-transparent (unacknowledged or acknowledged) RLC modes. On shared basic physical subchannels (SBPSCH, or PDCH), CS-1 to CS-4 as defined in R97 and MCS-1 to MCS-9 as defined in R99 normally apply.

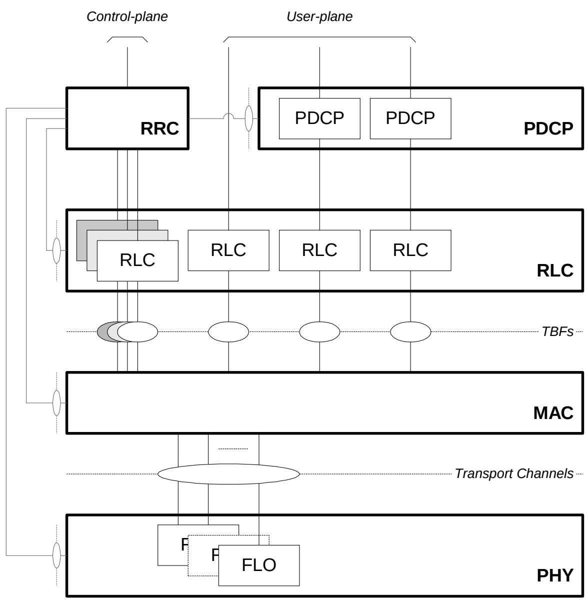

Figure 1. Protocol architecture for FLO in *Iu mode*

Temporary Block Flows (TBFs)
A TBF is a layer 2 logical connection used by two MAC entities to support the unidirectional transfer of RLC PDUs on basic physical sub-channels [5] [6].

Transport Channels (TrCH)
A transport channel is a channel offered by the physical layer to Layer 2 for data transport between peer layer one entities. A transport channel is defined by how and with which characteristics data is transferred on the physical layer. Given in GERAN FLO is available on DBPSCH only, only one type of transport channel is defined:

- Dedicated Channel (DCH): carries user or control data using GMSK or 8-PSK on a DBPSCH. A mobile station may have one or more transport channels of type DCH active at the same time in each direction.

Three different DCHs are defined in GERAN:

- UDCH: refers to a transport channel of type DCH used exclusively for carrying RLC/MAC blocks for data transfer belonging to *user-plane*;
- CDCH: refers to a transport channel of type DCH used exclusively for carrying RLC/MAC blocks for data transfer belonging to *control-plane*. The signalling TFC(s) (see [7]) shall be used when CDCH is active;
- ADCH: refers to a transport channel of type DCH used exclusively for carrying RLC/MAC blocks for RLC/MAC control message transfer. The signalling TFC(s) (see [7]) shall be used when ADCH is active.

On a transport channel, a transport block is the basic unit of traffic exchanged between the MAC sublayer and the physical layer. A transport block consists of a MAC PDU and contains exactly one RLC/MAC block.

Physical Layer
The physical layer of FLO is a simplified version of the layer one of UTRAN. Simplifications were possible because of the inherent characteristics of the physical layer of GSM/EDGE (for instance no spreading factor to take care of) and also because of the focus laid on the support of real-time services (no turbo codes selected). The architecture for FLO in GERAN, as depicted on Figure 2 below, includes CRC Attachment, Channel Coding, Rate Matching, Transport Channel Multiplexing, TFCI Mapping and Interleaving [8].

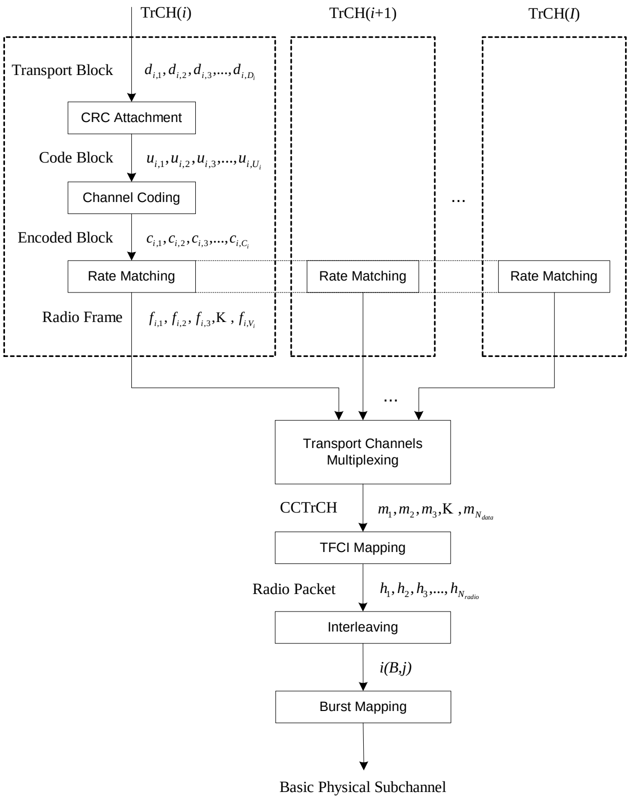

Figure 2. Physical layer architecture of FLO

On transport channels, transport blocks are exchanged between the MAC layer and the physical layer. The CRC attachment provides error detection for each transport block through a CRC. The size of the CRC to be used is a semi-static attribute, fixed on each transport channel and thus configured by Layer 3 to meet the QoS requirements of the service to be carried. For each transport block the CRC attachment provides one code block.

Code blocks are then processed in channel coding. The channel coding consists of a fixed mother code implemented as a non-systematic non-recursive convolutional code, the same code as in EGPRS.

After channel coding, the code blocks become encoded blocks and are treated in rate matching. The rate matching is the core of the FLO: not only it ensures that there are as many coded bits to be transmitted as there are bits available on the physical channel but also it balances the coding rate between different transport channel according to their relative importance. In rate matching, the bits of the encoded blocks are repeated or punctured according to the available bandwidth and according to the rate matching attributes (RMA). The RMAs define priorities between the coded bits of different transport channels, and are set by Layer 3 (semi-static attribute). For instance, by setting the RMA of a first transport channel to twice the value of the RMA of a second one, the coded bits of the first transport channel are made twice more important than the coded bits of the second one. This mechanism allows for balancing the coding rate between transport channels and therefore providing unequal error protection (UEP).

Outputs from the rate matching are called radio frames. For every radio packet the rate matching produces one radio frame per encoded block, i.e. per transport channel. Radio frames are then serially multiplexed into a Coded Composite Transport Channel (CCTrCH) by the transport channel multiplexing block.

After multiplexing the transport format combination indicator (TFCI) is inserted in order to specify the coding used for the current radio packet.

The coded TFCI and the CCTrCH are finally interleaved together on basic physical subchannel. The interleaving can be either diagonal or block rectangular and is configured at call set-up by layer 3.

VOIP OVER FLO

As previously described, the need for FLO is primarily driven by the introduction of real-time packet switched services. By taking VoIP as an example, this section shows how FLO improves the link level performance in contrast to EGPRS.

FLO Configuration

The speech frames (narrowband AMR codec) are transported by the RTP protocol. In order to conserve radio spectrum, the RTP/UDP/IP header is compressed at PDCP layer. The RLC protocol is running in unacknowledged mode and the MAC protocol in dedicated mode.

The error protection and detection is carried out in equal manner (EEP/EED). Hence, only one transport channel is needed. Each transport block contains a compressed RTP/UDP/IP header (28 bits), PDCP header (8 bits), RLC/MAC header (20 bits), and RTP payload. The RTP payload consists of CMR/TOC header (10 bits), RTP padding bits (0-7 bits), and speech bits (95-244 bits).

The size of the transport format combination indicator (TFCI) is set to four bits. The channel mode is limited to full-rate channels, but both modulation methods (GMSK and

8-PSK) are included. The error detection is carried out with a 12-bit CRC covering the header parts as well as the payload bits.

EGPRS configuration

The RLC payload contains the same fields as in FLO case, expect for the RLC/MAC header. The modulation and coding schemes are chosen so that that the selected MCS yields the maximum performance (C/Ico at FER=1%). It should be noted that such criterion does not always lead up to the minimum number of padding bits, since MCS 5 performs better than the MCSs 3 and 4. The selected modulation and coding schemes for the different codec modes are shown in Table 1. As can be seen, the number of padding bits for the GMSK modulated cases is 0-40, while 12-192 padding bits are needed with 8PSK.

Table 1. Modulation and coding schemes for VoIP over EGPRS

Codec Mode	RLC payload	Padding (GMSK)	MCS (GMSK)	Padding (8PSK)	MCS (8PSK)
12.2	296	0	3	152	5
10.2	256	40	3	192	5
7.4	200	24	2	24	2
5.9	164	12	1	12	1
4.75	142	34	1	34	1

Simulation Results

The simulations were performed on GSM900 band, the simulation length being 20000 speech frames. The results are summarized in Table 2, which shows the link level performance in terms of C/Ico at FER=1%.

Table 2. VoIP performance (C/Ico [dB] at FER=1%) in TU3iFH at 900MHz

Codec Mode	EGPRS (GMSK)	EGPRS (8PSK)	FLO (GMSK)	FLO (8PSK)
12.2	19.5	14.5	14.6	9.3
10.2	19.5	14.5	12.6	8.4
7.4	13.7	13.7	10.1	7.1
5.9	11.4	11.4	8.3	6.3
4.75	11.4	11.4	7.3	5.6

As can be seen, FLO improves the performance of the studied VoIP service from 3.1 to 6.9 dB with GMSK modes, and from 5.1 to 6.6 dB with 8PSK modes. It is interesting to note that even without any padding bits (AMR-12.2), FLO performs 4.9 dB better than EGPRS.

The main reasons for the bad performance of EGPRS are the granularity of RLC payload, short (20 ms) interleaving, and non-optimal RLC/MAC header. With FLO, the granularity of RLC/MAC payload is reduced to one bit and the interleaving depth is increased to 40 ms. In addition, the size of the RLC/MAC header is reduced to 20 bits, which is approximately half the size of the EGPRS header. While (E)GPRS coding schemes were designed for non real-time services on shared channels and provide optimal performance in such configurations, they were not originally designed for real-time services on dedicated channels.

CONCLUSION

This paper has shown the architecture and benefits of Flexible Layer One. In a nutshell, FLO allows the configuration of physical layer parameters at call setup, thus speeding up the introduction of new services and improving the link level performance in both circuit-switched and packet-switched domains.

The introduction of FLO is particularly important for the performance of real-time IMS services. As shown, FLO improves the link level performance of an AMR-based VoIP service from 3.1 to 6.9 dB compared to EGPRS. The main reasons for the improved performance are reduced granularity, longer interleaving, and smaller protocol overhead.

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