Applying List Output Viterbi Algorithms to a GSM-based Mobile Cellular Radio System

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ABSTRACT: Digital mobile radio systems based on the pan-European standards GSM and DCS 1800 are using concatenated coding schemes consisting of an inner error correcting convolutional code and an outer error detecting block code to ensure an appropriate performance. An improvement of this performance was achieved by applying list generating algorithms supplying a list of possible convolutional decoded sequences to the block decoder which checks them for correctness. Two list generating algorithms were investigated, namely the Serial List Output Viterbi Algorithm and the Soft Output Viterbi Algorithm with a novel list generating unit. Applying these algorithms yielded a gain of 3 dB without any bandwidth expansion nor additional delay. The only supplementary cost was a slightly increased computational effort. Therefore, List Output Algorithms seem to be a well-suited mean for improving the performance of GSM-based mobile radio systems. Concerning speech transmission, this improvement was achieved at the expense of a larger residual frame error rate because of the poor error-detecting capabilities of the used block code.

1 Introduction

Digital mobile radio systems combine many advantages over analogue networks such as the possibility of improved signal processing and more sophisticated signaling protocols. Since the early nineties, networks based on the pan-European standard GSM - Global System for Mobile Communication - are working at 900 MHz [1]. The underlying specifications have later been extended to include the frequency range of 1800 MHz (called DCS 1800). After overcoming children's diseases, these systems have become established.

Nevertheless, there are still many possibilities to improve the performance of such networks without changing the standards. Especially data transmission requires a very low bit error rate to keep re-transmissions within a certain limit and therefore avoiding an unnecessary high system load. The point of departure treated in this paper is the concatenated coding scheme consisting of an inner error correcting convolutional code and an outer error detecting block code [1]. Instead of using the conventional Viterbi Algorithm (VA) [2] for decoding the convolutional code, two List Output Algorithms have been implemented, namely the Serial List Output Viterbi Algorithm (SLVA) [5], [6], [7] and the Soft Output Viterbi Algorithm (SOVA) [3], [4] in combination with a novel list generating unit [8]. This unit takes into account both single errors and burst errors without an additional interleaver.

The advantage of such algorithms is based on the fact that the block decoder not only receives the most probable sequence from the VA, but also a list of the L most probable sequences. Therefore, the block decoder has the choice between L sequences within the list enlarging the probability to find the correct one. This modification does not require any bandwidth expansion nor any additional delay. The only supplementary cost is an increased computational effort.

This paper is organized as follows: Chapter 2 introduces briefly the relevant GSM-specifications concerning the coding procedure. Chapter 3 first treats some implementation aspects and then gives a short description of the SLVA and the SOVA combined with the new list generating algorithm [8]. Finally, chapter 4 discusses the simulation results for speech and data channels and chapter 5 concludes the main results.

2 Brief description of some GSM-specifications

Services defined by GSM can be principally divided into two groups: speech and data transmission. Concerning speech transmission, there exist standards for a full-rate and a half-rate channel using encoded data rates of 22.8 kbit/s and 11.4 kbit/s, respectively. Because of severe transmission conditions, unequal error protection is applied to the data stream as depicted in Figure 1 [1].

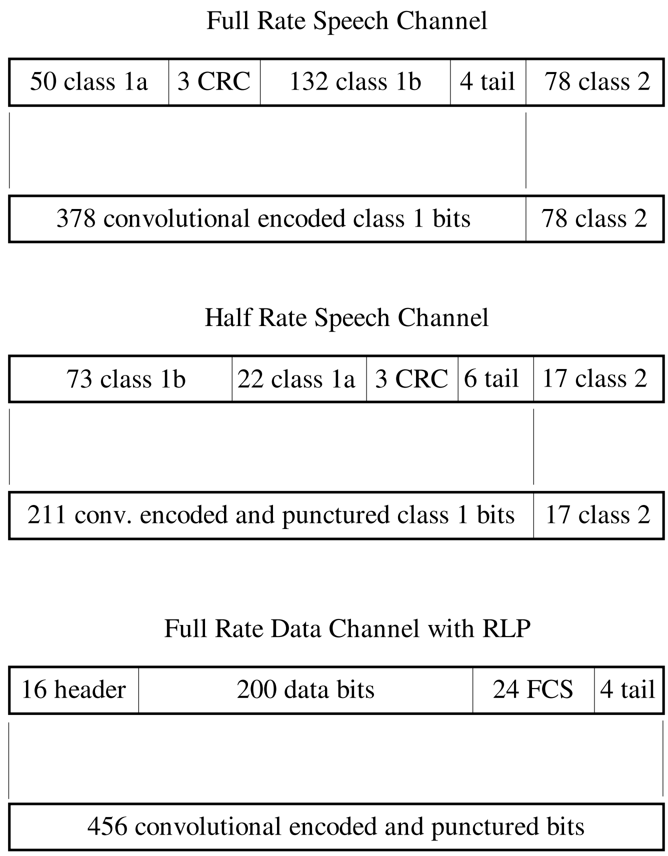

Figure 1: Unequal error protection for different GSM-channels

In detail, the most important bits, called class 1a bits, are first protected by a CRC (Cyclic Redundancy Check)-code adding 3 parity bits to the information data. Second, after appending the class 1b bits, a certain number of tail bits are added forcing the convolutional coder to return to the zero state at the end of each frame. In the case of a full-rate speech channel, all these bits are encoded by a half-rate convolutional code with memory $M = 4$ requiring 4 tail bits. The half-rate speech channel uses a convolutional code with a rate of 1/3 and a memory of $M = 6$. Here, puncturing shorten the encoded sequence to a length of 211 bits as depicted in Figure 1 (codewords associated with the 3 CRC-bits remain unpunctured) [10]. The most unimportant class 2 bits remain uncoded for both speech channels. A conventional Viterbi decoder [2] performs the decoding of the convolutional code. The CRC-decoder detects bad frames which are veiled so that no re-transmission is necessary.

Recently, high rate data transmission over mobile radio channels is playing a growing role. Here, the transparent and the nontransparent mode have to be distinguished. In both modes a convolutional code is applied as a forward error correction code (FEC) [1]. The two modes differ in the fact that the nontransparent mode additionally uses the radio link protocol (RLP), whereas the transparent mode is solely restricted to convolutional coding. The RLP consists of a header, the information sequence itself, and a parity check sum of it (see Figure 1). As in the case of speech transmission, the Viterbi algorithm performs the convolutional decoding. The frame check sequence (FCS)-decoder detects remaining errors. In the case of a transmission error, the frame has to be repeated.

3 List Output Algorithms

3.1 Introduction

Several researchers [5-7] have shown that list output decoding is a well-suited means for improving the performance of digital communication systems. One condition for applying these algorithms is using a concatenated coding scheme, such as the combination of CRC-codes and convolutional codes. The main idea behind this decoding strategy is to determine not only the most probable information sequence by the Viterbi algorithm (Maximum Likelihood Decoding), but to deliver a list of the L most probable sequences. Within this list the CRC-decoder chooses the correct sequence.

The procedure of checking the list-elements takes place as follows. In the case of an erroneous sequence, the CRC-decoder has to demand the next sequence from the list-output device. Therefore, signaling has to be done between the convolutional decoder and the CRC-decoder. For speech transmission, this signaling is no problem. In contrast to this, a serious problem arises concerning data transmission, where the two decoding devices take place at different sites in the uplink. The base transceiver station contains the convolutional decoder whereas the interworking function includes the CRC-decoder. In order to avoid a change of the existing standard allowing a more sophisticated signaling protocol between base transceiver station and interworking function, the decoding procedure has to be modified. Now, the base transceiver station decodes both the convolutional and the block code including the list generation. Then, the obtained sequence is re-encoded and transmitted to the interworking function. Hence, list generating algorithms are applicable in the downlink as well as in the uplink.

This paper compares different solutions concerning the determination of list elements. First of all, the next section briefly describes the optimal Serial List Output Viterbi Algorithm (SLVA). A more detailed description can be found in [5-7]. Second, section 3.3 discusses the SOVA with a new list generating unit.

3.2 Serial List Output Viterbi Algorithm

Based on the classical VA, the SLVA calculates first the most probable sequence called survivor. Within this procedure, the SLVA stores the cumulative metrics, the incremental metrics and the path history for every state and every time instance [7]. After determining the survivor, the second most probable sequence is computed exploiting the facts that it is associated with the second best cumulative metric and that it can deviate from the survivor only once [6]. Hence, the procedure for finding the lth best sequence is as follows:

1) Find the smallest metric difference between the l-1 previously sequences and all their discarded paths and save the re-merging points.
2) Calculate the cumulative metrics for all found paths at the end of the block for state 0...0.
3) The path corresponding with the smallest cumulative metric represents the lth best sequence and is sent to the output.
4) Exclude this path from the following search process by setting its metric difference to infinity.
5) Continue with 1): determine the next sequence by the same procedure until all L elements or the correct sequence have been found.

Therefore, the SLVA determines the L sequences within the list successively one after another. This procedure results in a very efficient algorithm because the next sequence is only calculated on demand if the block decoder indicates this.

3.3 SOVA with new List Generating Unit

In addition to the SLVA, the Soft Output Viterbi Algorithm in combination with a list generating unit has been investigated. A special algorithm was developed for the list generating unit taking into account both single errors and burst errors so that no special interleaver introducing additional transmission delay is necessary.

As in the case of the SLVA, the SOVA is an extension of the classical VA. In addition to the decoded data bits, it computes soft output values providing information of the reliability of the estimated data. A small amplitude indicates a large uncertainty of the corresponding bit. A more detailed description can be found in [3]. By means of these soft output values, a list generating unit estimates a list of possible sequences. In the past, two algorithms were introduced, namely the "1-state-method" and the "2^k-method" [6,7]. Both methods suffer because they need an additional interleaver spreading burst errors into single errors. This interleaver causes additional delay that is inadmissible especially for speech transmission. Furthermore, a supplementary interleaver would require a change of all existent devices, which is not desirable.

Therefore, a new list generating unit was developed that does not depend on an additional interleaver. It is a well-known fact that the VA produces both single errors and burst errors at its output. Hence, the new algorithm has to take care of both. It works as follows:

1) Determine the $L/2$-smallest reliability values.
2) Flip every bit associated with these values separately to detect $L/2$ single errors.
3) Flip two bits corresponding to the two lowest reliability values at once.
4) Continue this procedure until $L/2$ burst errors are determined.

Hence, this new algorithm provides a total of L list elements including the original sequence of the VA. For speech transmission, all mentioned list output algorithms have been modified in such a way that they consider only those bits protected by the CRC-code. Otherwise, alternative sequences deviating from the survivor at other positions than the protected ones could not be distinguished by the CRC-code. This would cause a degradation of the actual length of the list.

4 Simulation results

Simulations were carried out for the full-rate and half-rate speech channels and the full-rate data channel in nontransparent mode for a mobile radio system based on the GSM standard. The system model includes the complete frame structure of the GSM standard, the GMSK-modulation, the channel estimation by a training sequence and the equalization by the VA. Furthermore, the hilly terrain profile [9] was chosen as the channel model for all simulations. Besides the conventional VA as decoding algorithm the SLVA and the

SOVA with the new list generating unit described in section 3.3 were implemented.

4.1 Full-rate speech channel

Figure 2 illustrates the frame error rate P_f for the full-rate speech channel and different list generating units. As a reference, it also shows the frame error rate of the classical VA. Obviously, the SLVA with $L=4$ provides a gain of more than 3 dB at $P_f = 10^{-4}$. Increasing the number of list elements will not lead to further improvements (not shown in Figure 2, see section 4.2) so that 4 list elements are sufficient. The SOVA ensures gains of 2 dB for list length $L=4$ and $L=16$. As in the case of the SLVA, 4 list elements seems to be an adequate choice.

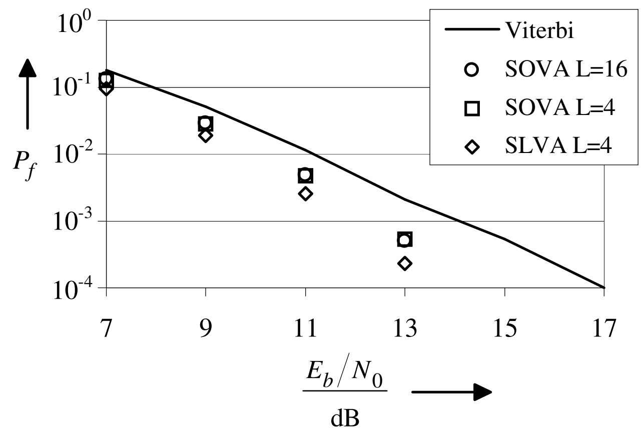

Figure 2: Frame error rate P_f for the full-rate speech channel

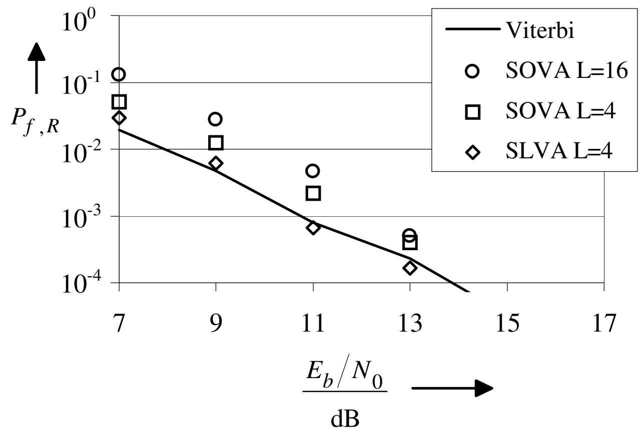

Figure 3: Residual frame error rate P_f for the full-rate speech channel

In opposition to these promising results, Figure 3 indicates that the residual frame error rate $P_{f,R}$ increases. $P_{f,R}$ represents the probability of errors that the CRC-decoder cannot detect because it receives a valid but wrong codeword. Implementing list output devices always enlarges the residual error probability because they only generate alternative sequences if the most probable sequence of the VA is detected as erroneous. In this case, these new sequences enlarge the probability of generating a valid but wrong codeword. Because of the poor error detecting capabilities of the CRC-code that uses only three parity bits, the residual frame error rates are quite high.

4.2 Half-rate speech channel

Concerning the half-rate speech channel, the SLVA yields nearly the same results (see Figure 4 and Figure 5). Here, even fewer information bits per frame are encoded so that the probability of producing a valid but wrong codeword by generating alternate list elements is growing. Additionally, Figure 4 illustrates that increasing the list length from $L=4$ to $L=16$ provides nearly no improvement (see Figure 5). For $L=16$ the residual frame error rate is even higher at small values of E_b/N_0 .

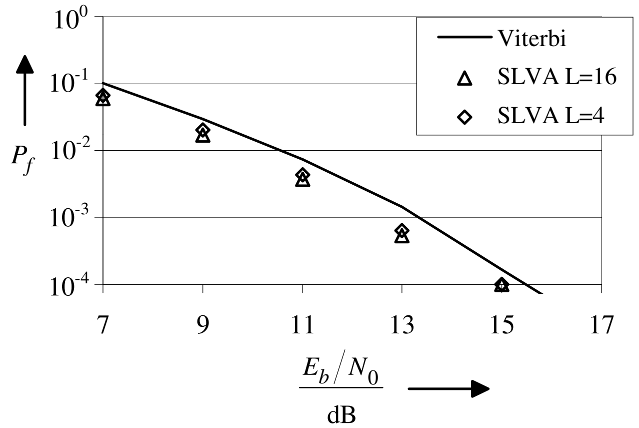

Figure 4: Frame error rate P_f for the half-rate speech channel

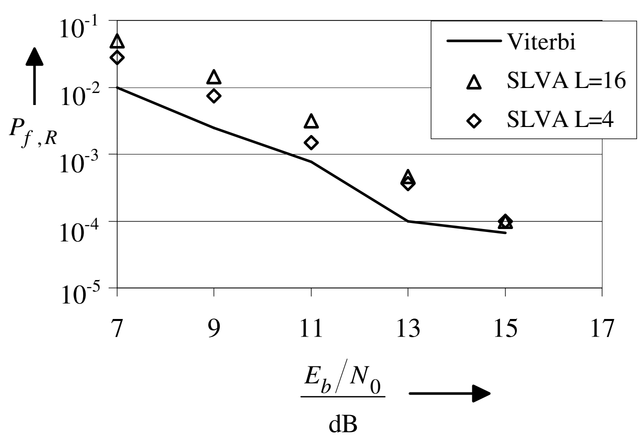

Figure 5: Residual frame error rate $P_{f,R}$ for the half-rate speech channel

4.3 Full-rate data channel

Figure 6 shows the obtained frame error rates for the full-rate data channel. It illustrates an increased performance of more than 2.5 dB for the SLVA with 16 list elements. Even 4 list elements provide a gain of 2 dB over the classical VA. Because of the remarkable extra costs for long list sizes, a compromise has to be made between an additional performance gain and the corresponding costs. Related to the presented simulation results, a list length of $L = 4$ is a good choice. The SOVA in combination with the new list generating unit performs only as well as the SLVA with $L = 2$. In contrast to speech transmission, the large frame check sequence of 24 parity bits ensures a residual error probability of approximately $P_{f,R} \approx 6 \cdot 10^{-8}$ which is much lower than the accuracy of the simulation. Therefore, no residual errors have been detected.

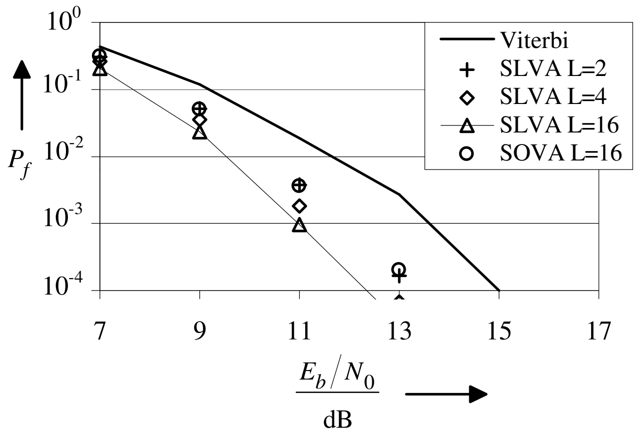

Figure 6: Frame error rate P_f for full-rate data channel

5 Conclusion

List Output Viterbi Algorithms are well suited for improving the performance of GSM-based mobile radio systems. The SLVA yields gains of more than 2.5 dB concerning data transmission. A compromise between computational effort and performance improvement produces a list length of $L=4$ that is an appropriate choice. A comparison of the SLVA with the combination of SOVA and new list generating unit showed that the SLVA always performs better. A valuation of the computational effort is very difficult because it depends strongly on the structure of error events concerning the SLVA. Taking into account the much better performance even for small list sizes, the SLVA should be preferred.

For speech transmission, the frame error rate decreased by one decade for the full-rate speech channel. Simultaneously, the residual frame error rate increased, enlarging the number of bad frames not being veiled. In order to answer the question if this effect is noticeable for the human listener, further investigations including speech codecs with have to be carried out. The new decoding scheme does not need any additional transmission delay nor any bandwidth expansion for the obtained improvement. Applying these algorithms in actual GSM- or DCS1800-systems will not even require a change of existing standards.

6 References

[1] Mouly, M., Pautet, M.-B.: *The GSM System for Mobile Communications*, 1992

[2] Forney, G.D.: "The Viterbi Algorithm", *Proceedings of the IEEE*, Vol. 61, No. 3, pp. 268-278, March 1973

[3] Hagenauer, J., Hoeher, P.: "A Viterbi Algorithm with Soft-Decision Output and its Applications", *Conf. Record IEEE GLOBECOM'89*, Dallas/Texas, Vol. 3, pp. 47.1.1-47.1.7, Nov. 1989

[4] Hagenauer, J.: "Source-Controlled Channel Decoding", *IEEE Transactions on Communications*, Vol. 43, No. 9, pp. 2449-2457, September 1995

[5] Seshadri, N., Sundberg, C.-E.W.: "Generalized Viterbi Algorithms for Error Detection with Convolutional Codes", *Conference Record IEEE GLOBECOM'89*, pp. 1534-1538, Dallas/Texas, Nov. 1989

[6] Seshadri, N., Sundberg, C.-E.W.: "List Viterbi Decoding Algorithms with Applications", *IEEE Transactions on Communications*, Vol. 42, No. 2/3/4, pp. 313-323, 1994

[7] Nill, C., Sundberg, C.-E.W.: "List and Soft Symbol Output Viterbi Algorithms: Extensions and Comparisons", *IEEE Transactions on Communications*, Vol. 43, No. 2/3/4, pp. 277-287, 1995

[8] Namokel, M.: *Implementierung und Beurteilung von List Output Viterbi Algorithmen*, University of Paderborn, March 1996, Dipl.-Ing. Thesis

[9] COST 207: *Digital land mobile radio communications*, Final report, Office for Official Publications of the European Communities, 1989

[10] Gerson, I., Jasiuk, M., Nowack, J., Winter, E.: „A proposal for speech and channel coding for the half-rate GSM channel“, *8. Aachener Kolloquium Signaltheorie*, pp. 33-36, 1994