Enhancing satellite system capacity using adaptive HARQ for delay tolerant services in mobile communications

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Abstract—We propose in this paper to introduce Adaptive HARQ in mobile satellite communication for delay tolerant services. Our motivation is that HARQ scheme, commonly used in terrestrial links, can be adapted to improve efficiency (in terms of throughput or system capacity) for mobile satellite communications. Our proposal uses the estimation of the Mutual Information of the received bits. We evaluate the performance of the proposed method with Land Mobile Satellite channel by means of simulations. Results are compared with those obtained with a static HARQ scheme. The adaptive retransmission technique we propose, shows a better performance in terms of efficiency while maintaining an acceptable delay for services.

Index Terms—Hybrid ARQ, Satellite Communications, Land Mobile Satellite (LMS) Channel, Delay, Efficiency.

I. INTRODUCTION

Link characteristics in mobile satellite communications make it difficult to exchange messages between transmitter and receiver(s). Long propagation delays (250 ms for geostationary satellite) is one of the main link characteristics in mobile satellite communications that can strongly affect the provided service. Another problem in mobile satellite communications are the transmission errors caused by the propagation impairments and the intra or inter-systems interference.

Our objective is to propose a mechanism enabling to reach the best use of the bandwidth and thus improve the efficiency of link usage while providing an appropriate service to applications. The targeted services (data transfer from sensors, messages for aeronautical services ...) are assumed to be tolerant to delay. For example some aeronautical services define delay requirement for the delivery of 95% of messages.

In this paper we propose to introduce adaptivity in HARQ mechanism. Hybrid ARQ (HARQ) protocols are used in most of recent terrestrial wireless communication systems. Static HARQ, which transmits a fixed number of bits at each transmission, is not optimal from the efficiency point of view since sometimes the transmitter transmits some extra parity bits more than needed to decode the message. However, adaptive retransmissions can improve the system performance and throughput level. Many papers have studied adaptive retransmissions using Adaptive Coding Modulation (ACM) and HARQ. Other contributions proposed combinations of ACM or HARQ, where soft combining of transmission blocks in the receiver and block flat-fading channel are assumed [1].

In [2] an approach based on the channel states for adaptive coding and modulation for mobile satellite communications is presented. Also ACM was studied in [3] with a multi-layer coding (MLC) in the forward link and open-loop adaptation in the return link. In [4] a long study about channel estimation and physical layer adaptation techniques was made. Adaptive HARQ was studied also in [5] for a scheme based on punctured LDPC codes.

The main idea of our adaptive technique is to send a feedback to the transmitter containing the number of bits needed to decode the packet with a targeted probability, if it has not been decoded successfully during the previous transmissions. This technique uses the mutual information of the received bits to predict the number of bits needed to decode at a predefined decoding probability for each transmission. It uses the knowledge of the statistical distribution of attenuation in the channel. Some adaptive retransmission techniques based on mutual information have been studied [6] [7], but not in a satellite communications environment neither using mutual information in combination with predefined decoding probability control. Our adaptive HARQ transmission proposal is simulated in a satellite communications environment, where Land Mobile Satellite (LMS) channel and long Round-Trip time are considered.

The remainder of this paper is organised as follows. Section II describes channel capacity and Land Mobile Satellite channel model. Two techniques of transmission using HARQ are presented in Section III. The results of simulations and the
comparison between both techniques are presented in Section IV. We conclude our study in Section V.

II. CHANNEL MODELLING

In our study on mobile satellite communications, we considered the Land Mobile Satellite (LMS) Channel to simulate this environment [8] [9]. In the following we present the channel capacity and model, where these two characteristics serve to simulate transmission scenarios as described in Section III.

A. Channel capacity and Mutual Information

Channel capacity $C$ quantifies the maximum achievable transmission rate of a system communicating over a band-limited channel, while maintaining an arbitrarily low error probability. It corresponds to the maximum of the mutual information between the input and output of the channel, where the maximization is done with respect to the input distribution. Mutual information measures the information that $X$ (input of the channel) and $Y$ (output of the channel) share. It is a key parameter in our approach, as it will be used to calculate the number of bits needed to decode a message at the next transmission with a given probability. For an equally distributed input probability, the mutual information, which corresponds to the capacity, can be calculated by the following equation [10]:

$$MI\left( \frac{E_s}{N_0} \right) = \log_2(M) - \frac{1}{M(\sqrt{\pi})^N} \times \sum_{m=1}^{M} \int_{-\infty}^{+\infty} \int_{-\infty}^{+\infty} \exp\left(-|t|^2\right) \times \log_2 \left[ \sum_{i=1}^{M} \exp\left(-2t.d_{mi} - |d_{mi}|^2\right) \right] \ dt \ (\text{bits/symb})$$

where:
- $M$ is the modulation order.
- $N$ is the space dimension, depends on the used modulation ($N=2$ for any PSK-based modulation with more than 2 states).
- $t$ is the integration variable ($t = (t[1],...,t[N])$).
- $d_{mi} = \sqrt{\frac{E_s}{N_0}}(x_m - x_i)$ ($x_i$ is an input symbol).

For the rest of the paper, we define $MI_{req}$ as the average MI per bit required to decode a packet at a given probability expressed in Packet Error Rate (PER). $MI_{req}$ can be calculated using the mutual information function [1] and the performance curves giving the PER versus $E_s/N_0$ of the used modulation/coding scheme [11]. For a given PER, we take the corresponding $E_s/N_0$ and we use it with the mutual information function to obtain the mutual information required to decode a codeword at this PER. The prediction of performance of the Packet Error Rate (PER) based on MI is quite classical, and has been described and validated in [12].

Given the input symbol $x_i$ and its energy $E_{si}$ and a realisation of noise $n_i$ (has a Gaussian distribution with variance $N_0$), assuming perfect knowledge of the channel attenuation $\rho_i$, the output symbol can be written as:

$$y_i = \rho_i \sqrt{E_{si}}x_i + n_i$$

From now on, to compute the mutual information we use [1], which takes as input $\rho^2 \frac{E_s}{N_0}$, where $\frac{E_s}{N_0}$ is the average energy per symbol.

B. LMS Channel

There are many differences between the propagation for a terrestrial link and for a satellite link. A reference propagation model for the Land Mobile Satellite (LMS) Channel is a statistical model based on a three state Markov chain [8]. This model considers that the received signal originates from the sum of two components: the direct signal and the diffuse multipath. The direct signal is assumed to be log-normally distributed with mean $\alpha$ (decibel relative to LOS (Line Of Sight)) and standard deviation $\Psi$ (dB), while the multipath component follows a Rayleigh distribution characterized by its average power, $MP$ (decibel relative to LOS). This model is called Loo distribution [13] [9]. For the modelling of the LMS channel in our simulations, we use propagation time series using a propagation simulator based on the three state channel [8] [9] provided by CNES. Using this tool we calculate the distribution of the probability to obtain an attenuation in the channel for a given environment.

III. DESCRIPTION OF THE HARQ SCHEMES

This Section presents the two different techniques of transmission taken into consideration in our study. Both techniques use HARQ, precisely Selective Repeat (SR) ARQ Type-II (Incremental Redundancy). But before presenting these two techniques, we present the different assumptions taken into consideration.

A. Assumptions

- The synchronisation is never lost.
- The transmitted codewords (useful or parity bits) are identified with a sequence number that is never lost.
- The return channel does not introduce errors and the feedback can be transmitted immediately (no congestion problem on the reverse link).
- We suppose the pre-knowledge of the global statistics of the channel (for the case of an adaptive approach).

In the following we refer to “packet” as the cumulative bits transmitted or retransmitted, at each step of the process, until the decoding of a given codeword, or reaching the maximum number of bits per codeword.
B. Static (Non-Adaptive) HARQ

Static HARQ can be described as follows. The sender transmits a number of bits (data bits + parity bits) that correspond to a given packet. After receiving the Feedback (ACK/NACK) from the receiver, the transmitter decides to no longer send bits corresponding to this packet if an ACK is received, or to send more parity bits if a NACK is received. The number of bits to be sent in the next retransmission is determined according to a table predefined at the sender. This table is designed according to the HARQ Incremental Redundancy (Type II), where the number of bits to be transmitted at each transmission is predefined at the beginning of the communication without any pre-knowledge about the channel quality. If the cumulative received sequence can not be decoded after a maximum number of bits transmitted, the transmitter stop sending bits that correspond to this packet. The parity bits are generated according to a coding scheme with a code rate corresponding to the maximum number of bits that can be transmitted per codeword. The bits to be sent at each transmission are part of the original codeword (mother code), leading to a different code rate at each transmission (see Fig. 1). This technique of transmission is somehow similar to puncturing. The data bits of each packet must be kept at the buffer of the sender as long as the packet is still not decoded or the transmitter has not decided to end the transmission. At the receiver side, the LLRs (the Likelihood Ratio) of received bits of each packet must be kept in the buffer as long as the packet is not decoded or the transmission of the corresponding bits has not ended.

As mentioned before, there is a fixed number of bits to be transmitted at each transmission. $N_{b_j}$ refers to the number of bits to be transmitted at the $j^{th}$ transmission.

C. Adaptive HARQ

1) Adaptive Retransmission Model:

We now propose a method called “HARQ Adaptive Retransmission using Mutual Information”. Our method differs from the non-adaptive HARQ by the feedback which contains an estimation of the number of bits still required for the receiver to decode the packet if it is not decoded from the previous transmissions. So, the receiver computes the number of bits needed to decode the packet directly in the next retransmission, and sends that number in the feedback to the transmitter.

Our proposal is to compute the number of extra parity bits needed when the codeword cannot be decoded, using Mutual Information (MI). After receiving some bits of a given packet, the receiver calculates the MI accumulated for this packet. Considering a reference $E_s/N_0$, which is a fixed value of $E_s/N_0$ in clear sky (for a given terminal and without attenuation). We suppose that the receiver can estimate the channel quality at this moment of transmission and assuming a perfect estimation. The model assumes that the channel is stationary for the transmission time of bits, at a given transmission for a given packet. The MI obtained at the $j^{th}$ transmission for a given packet can be computed as :

$$MI^{(j)} = N_{s_{\text{sent}}}^{(j)} \cdot MI((\rho^{(j)})^2 \cdot \frac{E_s}{N_0})$$

The MI per bit accumulated for a given packet, from the beginning of transmission until the $j^{th}$ transmission, can be computed as :

$$MI_{\text{acc}}^{(j)} = \frac{N^{(j-1)} \cdot MI_{\text{acc}}^{(j-1)} + MI^{(j)}}{N^{(j)}}$$

where :
- $\rho^{(j)}$ is the attenuation coefficient affecting bits transmitted at the $j^{th}$ transmission for a given packet.$(\rho^{(j)})^2$ is estimated by receiver measurement.
- $N_{\text{sent}}^{(j)}$ is the number of bits transmitted at the $j^{th}$ transmission, affected by $\rho^{(j)}$.
- $MI(\cdot)$ is the mutual information function.
- $N^{(j)}$ is the total number of bits transmitted for a packet up to the $j^{th}$ transmission.
- $MI_{\text{acc}}^{(j)} = 0$.

Let us consider $MI_{\text{acc}}^{(j+1)}$ the MI per bit considered for the next transmission. $MI_{\text{acc}}^{(j+1)}$ is the key parameter for the prediction of the number of bits to be sent in the next transmission. We will explain the method to choose this value later.

Then the number of bits $N_{\text{req}}^{(j+1)}$ to be transmitted at the next transmission can be obtained by the following equation:

$$N_{\text{bits}} \cdot MI_{\text{req}} = N^{(j)} \cdot MI_{\text{acc}}^{(j+1)} + N_{\text{req}}^{(j+1)} \cdot MI_{\text{acc}}^{(j+1)}$$

Note that $N_{\text{bits}}$ is the maximum number of bits that can be transmitted for a packet.

Finally $N_{\text{req}}^{(j+1)}$ is given by :

$$N_{\text{req}}^{(j+1)} = \frac{N_{\text{bits}} \cdot MI_{\text{req}} - N^{(j)} \cdot MI_{\text{acc}}^{(j+1)}}{MI_{\text{acc}}^{(j+1)}}$$

Note that this technique has some similarities with models studied in [6] [7] which are based also on Mutual Information. However these models do not use decoding probability control as detailed in the next section.

2) Decoding probability control: The Land Mobile Satellite Channel is a channel that changes quickly. Moreover, the round trip time in satellite communications is long (~500 ms for geostationary satellites). So even if the receiver can estimate the channel quality for the last sequence of received bits, this information will not be useful at the transmitter considering the long delay and the highly variable channel.

Our proposal is based on the mutual information where the ideal value of $N_{\text{req}}^{(j+1)}$ is computed from $MI_{\text{acc}}^{(j)}$ and $MI_{\text{acc}}^{(j+1)}$.
which depends on the channel quality at the next transmission. Since this last value can not be known, we use the knowledge of the statistical distribution of channel attenuation to control the probability of decoding for a packet at each transmission.

The idea is to define at the beginning of the communication, a table containing the probability of decoding at each transmission. These probabilities are chosen according to delay constraints of the application, keeping in mind that the decoding probability and the efficiency are related. The sender can transmit a large number of bits at the first transmission, which increases the decoding probability but the efficiency will decrease and vice versa. So we have to improve efficiency while respecting delay constraints for services. In addition the sender may also want to limit the number of transmission attempts for a given codeword. Note that after the maximum allowed retransmission steps, the buffered LLRs are discarded.

In the rest of this paper we will consider \( P_j \) the probability of decoding at the \( j^{th} \) transmission, where \( P = \sum_j P_j \) is the percentage of decoded packets over all the transmitted packets.

To target a decoding probability \( P_j \) at the \( j^{th} \) Transmission, the receiver has to find the corresponding \( M_{N_{next}}^{(j)} \) necessary to calculate the number of bits \( N_{needed}^{(j)} \) to be transmitted

\[
M_{\text{next}}^{(j)} = M\left(\left(\rho_{\text{next}}^{(j)}\right)^2 \frac{E_s}{N_0}\right) \tag{7}
\]

As we noticed before, \( M\left(\cdot\right) \) is the mutual information function that takes as input \( \left(\rho_{\text{next}}^{(j)}\right)^2 \frac{E_s}{N_0} \)

\( M_{\text{next}}^{(j)} \) is pre-computed, it depends on the \( j^{th} \) element in the probability decoding table and also on \( \sum_{k=1}^{j-1} P_k \) as we will see later in (9). We assume reference \( E_s/N_0 \) fixed for a given terminal (only \( \rho^2 \) changing over the time). So, \( M_{\text{next}}^{(j)} \) depends only on the channel attenuation \( \rho_{\text{next}}^{(j)} \). The mutual information \( M\left(\cdot\right) \) is a strictly increasing function (as a function of \( \rho \)). Then any attenuation coefficient greater than \( \rho_{\text{next}}^{(j)} \) will lead to a successful decoding. To determine \( \rho_{\text{next}}^{(j)} \) leading to \( P_j \), we use the Cumulative Distribution Function (CDF) of LMS Channel.

To simplify our calculation, we will consider these two events:

- \( A_j \) : Successful decoding at the \( j^{th} \) transmission.
- \( B_{j-1} \) : Not decoding at the \( (j-1)^{th} \) transmission.

\( P_j \) can be defined as \( p(A_j \cap B_{j-1}) \) and \( p_j \) is \( p(A_j) \). Since \( A_j \) and \( B_{j-1} \) are independent (according to the channel modelling),

\[
P_j = p_j (1 - \sum_{k=1}^{j-1} P_k)
\]

\[
p_j = \frac{P_j}{1 - \sum_{k=1}^{j-1} P_k}
\]

(8)

Note that the CDF of the channel gives us \( P(\rho \leq \rho_j) \). On the other hand, \( p_j \) corresponds to \( P(\rho \geq \rho_j) \). Therefore, \( \rho_{\text{next}}^{(j)} \) on the CDF graph is given by \( 1 - p_j \) (See Fig. 2).

Once we found \( \rho_{\text{next}}^{(j)} \) leading to \( P_j \), we use it in (7). Then we use \( M_{\text{next}}^{(j)} \) in (6) to calculate \( N_{\text{needed}}^{(j)} \), necessary to obtain a decoding probability \( P_j \) at the \( j^{th} \) transmission.

Fig. 2: The cumulative distribution function of the attenuation coefficients in LMS Channel

IV. SIMULATIONS AND RESULTS

We present the results obtained by implementing both schemes (non-adaptive and adaptive HARQ) described in the previous Section, and we compare these results.

To be fair, we propose a model (derived from the non-adaptive HARQ) which allows us to compare these two methods. However, we define a set of precomputed decoding probabilities, that provides for each round of retransmission a precomputed number of parity bits to send. We are interested to improve the efficiency while maintaining an acceptable delay for services. So the idea is to set the decoding probabilities to the same values for both models (Adaptive and Non-Adaptive), and then compare the resulting efficiencies.

To obtain the fixed decoding probabilities at each transmission, the mean number of bits to be sent should be pre-calculated. This pre-calculation is done using (4) and (6), where \( M_{\text{next}}^{(j+1)} \) is the necessary mutual information to obtain a given decoding probability at a fixed reference \( E_s/N_0 \) and \( M_{\text{acc}}^{(j)} \) is the mean mutual information per bit accumulated for all packets that could not be decoded at the previous transmissions. Both of them are calculated using the channel distribution and decoding probability at each transmission. \( N_{\text{sent}}^{(j)} \) in (3) is the mean number of bits already calculated for the \( j^{th} \) transmission using (6). Once the number of bits to be transmitted at each transmission is calculated, they must be put in a predefined table and used as in the first scenario (non-Adaptive HARQ).

The system parameters that we will consider are:

- Satellite Orbit : Geostationary (GEO)
- Round Trip Time : 500 ms
- Band : S
- Land Mobile Satellite Channel , Intermediate Tree Shadowed Environment (ITS)
- Speed : 60 Km/h
- Distance : 10 Km
- Mother FEC code, CCSDS Turbo Codes 1/6
- Codeword Length : 53520
- Modulation : QPSK
- Symbol Time : $4 \times 10^{-6}$ seconds, bit rate (Rb) : 500 Kbps
- Our simulations are about 10 minutes of communication between the transmitter and the receiver (about 300 Mb transmitted)

In our simulations we consider a targeted PER of $10^{-4}$, and we use the actual performances of CCSDS Turbo Codes (8920, 7) as presented in [11]. The simulations use the computation of mutual information, as a real receiver would implement, according to the previously proposed algorithm. Doing this have required (in simulations) a calibration phase that has taken into account the actual numerical performances of the targeted FEC code(s). This allows to avoid implementing a real decoder in the simulation chain, while assuring a very good accuracy of the representation [12]. To simplify, we use the mutual information to decide if the packet is decoded or not using this formula:

$$N^{(j)} MI_{acc}^{(j)} \geq N_{bits} MI_{req}$$

(9)

If the number maximum of bits allowed (codeword length) is reached without successfully decoding the packet, the transmitter stops the transmission that corresponds to this packet. A configurable maximum number of transmissions $N_{max}$ is allowed in our simulations for each message, then the sender starts transmitting the next message. In the simulations, non decoded sent codewords provide no contribution to efficiency.

Finally, we define the efficiency, $E$ (bits/symbol), as follows:

$$E = \frac{N_{data} \times N_{data}}{N_{total}}$$

(10)

where

- $N_{data}$ is the number of data bits (useful bits) per codeword considered in our coding scheme,
- $N_{decoded packets}$ is the total number of decoded packets during the communication,
- $N_{total}$ is the total number of symbols transmitted during the communication.

In the following we have considered three cases for our adaptive HARQ approach (explained in the previous Section). In Case 1, only one transmission is allowed. In Case 2, two transmissions are allowed and in Case 3, four transmissions are allowed. Their decoding probability are given in Table I.

Note that these probabilities do not represent values for specific applications. They are chosen arbitrarily to test our model. The last value in the table is 0.9999 and not 1 because PER considered in our simulations is $10^{-4}$, so the total probability of decoding is $1 - 10^{-4}$.

<table>
<thead>
<tr>
<th>Transmission</th>
<th>$P_1$ (Case 1)</th>
<th>$P_2$ (Case 2)</th>
<th>$P_2$ (Case 3)</th>
<th>$P_4$ (Case 4)</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_1$ (Case 1)</td>
<td>0.9999</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>$P_1$ (Case 2)</td>
<td>0.5</td>
<td>0.9999</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>$P_1$ (Case 3)</td>
<td>0.5</td>
<td>0.3</td>
<td>0.1</td>
<td>0.0999</td>
</tr>
</tbody>
</table>

TABLE I: Predefined decoding probability table for 3 cases considered

Fig. 3 presents the efficiency obtained after simulating these three cases for different values of reference $E_s/N_0$ (NB: the channel variations $\rho^2(t)$ still being applied to this reference $E_s/N_0$, as explained in previous Sections). This figure shows that case 3 (allowing four transmissions) outperforms case 1 and case 2 (allowing only one transmission and two transmissions respectively) in term of efficiency. This result can be interpreted by the fact that allowing only one transmission requires a large number of bits to be transmitted, sometimes more than the number of bits needed, to insure a complete decoding. When many transmissions are allowed, the number of bits to be transmitted at each transmission is optimized and computed in an adaptive way which optimizes the efficiency. This shows the importance of allowing many transmissions for the same packet.

Fig. 3: Efficiency obtained with three cases considered for different values of reference $(E_s/N_0)dB$

Our target is to optimize the efficiency while respecting delay constraints for a given service. One of the solutions is to improve the efficiency per retransmission phase, while fixing the decoding probabilities for both models (adaptive and non-adaptive). Thus we figure out the gain on efficiency of our adaptive approach against the non-adaptive model based on fixed table of number of bits.

In the following we have considered adaptive and non-adaptive techniques. Fixed decoding probabilities at each transmission considered in our simulations are given in Table I (case 3) where the maximum number of transmissions for a given packet is four. It corresponds to a service accepting the delivery of 80% of the messages at the first two transmissions and 20% at the last two retransmissions.

After simulating first (non-adaptive HARQ) and second (Adaptive Approach) scenarios, under the same conditions, we obtain the same decoding probabilities per retransmission phase for both models as we wanted. In fact, we need to obtain same decoding probabilities in order to compare the efficiency. In the following, we express the delay for decoded packets (at the receiver) in terms of number of transmissions ($N_{trans}$), bit rate ($R_b$), number of bits sent ($N$) and propagation delay ($T_{propag}$), assuming a negligible access delay:

$$\text{Delay} = \frac{N}{R_b} + 2(N_{trans} - 1)T_{propag} + T_{propag} \text{ (s)}$$

(11)
Fig. 4 shows the mean delay (right axis), required to decode packets, obtained with both models (by means of simulations). The mean delay is computed by averaging delays obtained for decoded packets calculated using $\frac{E_s}{N_0}$. As we can see, the values obtained are approximately the same. The delay stays stable and controlled by the decoding probability, globally constant. It changes a little, from a reference $E_s/N_0$ to another, according to the number of bits transmitted and the values of decoding probabilities obtained which are approximately the same as in Table I (case 3). To show the performance of the proposed Adaptive model, Fig. 4 compares also the efficiency (left axis) obtained with both models in the same conditions and same decoding probabilities (same delay). We can see that the Adaptive model outperforms the static one (Non-Adaptive) especially for high values of reference $E_s/N_0$, with at least a gain of 8%. These results seem promising, however has been obtained without any optimisation of decoding probability for each transmission step. Therefore, some further improvements could probably be obtained. This clearly calls for an optimization process in further steps of the work.

As a future work, real system parameters must be considered at the physical layer (in particular framing and overhead) to evaluate the performance of the mechanism in real systems. We will focus also on the decoding probabilities and the maximum number of transmissions allowed to find optimal values in order to reach good tradeoff, or to maximize efficiency while satisfying delay requirements that can be expressed by some services. Finally, we plan to study the effect of this approach on the higher layers.

V. CONCLUSION

In this paper, we have studied two techniques of HARQ transmission for satellite communications. The first one is a non-adaptive scheme with any pre-knowledge of the channel; the second one is an adaptive technique which takes into account the channel quality at each transmission. This adaptive technique relies on Mutual Information to predict the number of additional bits needed to decode a packet that could not have been decoded from the previous transmissions. This prediction based on the Mutual Information is combined with a method to control the decoding probability at each transmission knowing the channel distribution. Finally, results obtained after simulating both scenarios in a mobile satellite communication environment are compared in terms of decoding probability (delay) and efficiency. Results show that the adaptive scenario has better performance especially for high values of reference Signal to Noise Ratio.

REFERENCES


