



SDI Review Form 1.6

Journal Name:	British Journal of Mathematics & Computer Science
Manuscript Number:	2013_BJMCS_6547
Title of the Manuscript:	Audio CODECs using the interpolation "LSM" and comparison with PCM
Type of the Article	Short Communication

General guideline for Peer Review process:

This journal's peer review policy states that **NO** manuscript should be rejected only on the basis of '**lack of Novelty**', provided the manuscript is scientifically robust and technically sound.

To know the complete guideline for Peer Review process, reviewers are requested to visit this link:

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PART 1: Review Comments

	Reviewer's comment	Author's comment <i>(if agreed with reviewer, correct the manuscript and highlight that part in the manuscript. It is mandatory that authors should write his/her feedback here)</i>
Compulsory REVISION comments	1. A number of citations are given that present related work. However, related work is not thoroughly detailed. I suggest adding more details about similar coding methods as well as commenting on these methods' results. 2. Section 2: The authors state that they used polynomial models to represent the audio because LSM with polynomials have shown to provide the best fitting. Provide evidence or references that this is the case. Also provide and explain in more detail other works that considered LSM and polynomials for audio coding. 3. Section 3 and 4 are a bit confusing. Consider changing the title of Section 3 to "proposed algorithms" and include all your experimental results to section 4. 4. The details of each algorithm are stated nowhere in the paper and your description on the tables has some major missing parts which make all the algorithms non-comprehensive. For example: a) Alg 2 decoding part: step 2: Is there a word "samples" missing? b) Alg 2 decoding part: step 3 should be here? Isn't it part of the encoding phase? c) Alg. 3 encoding part: From input y is missing d) Alg. 3 step 2. Again the variable y is missing Include in Section 4 some text that describes the algorithms properly.	Regarding (1): We provided some of the approaches found in the literature, just to show our awareness about the context. We clarified more that we are focusing on replacing the quantization part in CODECs/Modulations techniques. We showed now detailed explanation about how this works and we applied our algorithm to MP3 files. Regarding (2-6): We rearranged the paper totally. We put flowcharts and detailed explanation about the steps and the experiments and the results. All the algorithms rewritten in a sufficient way and we fixed all the typos in the Pseudocodes. Regarding (7): Our research aims to open new prospective for using some of the known interpolation methods to improve audio processing. We are aiming to publish our research that focuses on the methodology and the results without discussing the complexity. The complexity of least squares is well presented in many references, but we didn't work on that part. There are tens of researches that didn't discuss the complexity and just leave the door opened for other interested researchers to discuss that based on their areas of expertise.



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	5. In the second algorithm in step 2 of encoding you perform a search to find the Ns which adaptively changes. But the paper does not describe how this search is performed algorithmically. For example, does it start from 1 sample and increases by one in each iteration? What is the stop criterion is not met after a big number of iterations? Is this possible to happen? For all these questions that a reader will have when reading the algorithms, I strongly suggest a complete section with all the algorithmic details in depth. 6. Algorithm 3 is suggested in order to increase convergence. But this is not obvious how is done and the paper provides no details. You could describe the intuition and motivation for this algorithm and also calculate the convergence of each algorithm and compare them in a figure. 7. In addition to comment (6) the computational efficiency of your algorithms needs to be stated. Are the proposed algorithms slow or fast compared to other audio coding approaches? You could compare the computational efficiency--in terms of processing time or complexity---of your algorithms and report it in your research. 8. The results seems promising, however SNR and goodness of fit can only give some objective measures of how good a coding method is, but they are not correlated with subjective measures in a 100%. As the perceptual quality of the coded audio is the most important aspect in audio coding, I would consider a listening test mandatory for your research. Listening tests is the most common and valid way in the audio processing and coding community to validate the	Regarding (8-9): some of the methods mention that their efficiency can get a range of SNR between "a" and "b". That is why we preferred to show that in many parts we reach very high maximum SNR even better than PCM quantization part.
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	efficiency of a method. You could refer to the ITU methodology for ABX tests to help you conduct your listening tests according to the community standards. 9. It is not obvious to me what you mean by Max. SNR in your figures. The maximum SNR is for sure the most optimistic case and not very representative measure. You could consider the mean segmental SNR.	
<u>Minor</u> REVISION comments	You could replace the figures with tables and present results of SNR and r% separately, as it would be easier to read and interpret the results this way. I think the information in each figure is very much now, which makes it a bit hard to read.	
<u>Optional/General</u> comments	Throughout the paper, I also noticed some language deficiencies that make your work a bit hard to understand.	We reviewed the paper with New Mexico Tech writing centre and modified all language mistakes.