Patents Act 1977 | **Opinion Number** | **21/21**

OPINION UNDER SECTION 74A

Patent	EP 3132443 B1
Proprietor(s)	VoiceAge Corporation
Exclusive Licensee	
Requester	Jenner & Block London LLP
Observer(s)	VoiceAge Corporation (represented by Mathys & Squire LLP)
Date Opinion issued	10 January 2022

The request

1. The comptroller has been requested to issue an opinion on the validity of at least claims 1 and 10 of European Patent (UK) 3132443 B1 (the patent) in the light of the following prior publications:

   Jean-Marc Valin, Roch Levebvre, "Bandwidth Extension of Narrowband Speech for Low Bit-Rate Wideband Coding," Proc. IEEE Speech Coding Workshop (SCW), 2000, pp. 130-132, February 2000, DOI:10.1109/SCFT.2000.878425 ("**VALIN 1**")

   Jean-Marc Valin, "Extension spectrale d'un signal de parole de la bande téléphonique à la bande AM", University of Sherbrooke, December 2001 ("**VALIN 2**")

   J. D. Markel, A. H. Gray, "Linear Prediction of Speech," Springer-Verlag, Berlin Heidelberg, New York, 1976, ISBN 3-540-07563-1, pages I-XII and 129-163, published 1976 ("**MARKEL-GRAY**")

   John Makhoul, "Linear Prediction: A Tutorial Review," Proc. IEEE, Vol. 63, pages 561-580, April 1975, published April 1975 ("**MAKHOUL 1**")

   John Makhoul, "Spectral Linear Prediction: Properties and Applications," IEEE Trans. Acoust., Speech, Signal Processing, Vol. ASSP-23, No. 3, pp. 283-296, June 1975, published June 1975 ("**MAKHOUL 2**")

   P.P. Vaidyanathan, "The Theory of Linear Prediction", Morgan & Claypool Publishers, 2008 ("**VAIDYANATHAN**")

Observations

2. Observations were received on 18 November 2021 including a copy of a preliminary opinion dated 21 June 2021 issued by the German Federal Patent Court regarding the patentability of the patent. Observations in reply were received on 1 December 2021 including a further document:

   Gernot Kubin, W. Bastiaan Klejin, "Speech Watermarking for Analog Flat-Fading Bandpass Channels", IEEE Transactions on Audio Speech and Language Processing, December 2009 ("**KUBIN**")

3. The requester states that this document has been cited to further demonstrate the common general knowledge of the person skilled in the art (VAIDYANATHAN being cited originally as an example of this common general knowledge). Therefore, I consider that this document is being used to further the request, rather than show flaws in the observations, and so I will not consider this document. It is fundamental to the opinions process that the requester raises their best argument at the outset. This allows the observer an opportunity to comment on the entire argument. If argument is introduced at the observations-in-reply stage, then the observer is not able to respond and so such argument is not allowed.

Matters to be considered by this Opinion

4. Section 74A of the Patents Act provides for the procedure where the Comptroller can issue, on request, non-binding opinions on questions of validity relating to novelty and inventive step.

5. Section 74A(3) of the Patents Act 1977 states:

   The comptroller shall issue an opinion if requested to do so under subsection (1) above, but shall not do so –

   (a) in such circumstances as may be prescribed, or

   (b) if for any reason he considers it inappropriate in all the circumstances to do so.

6. Rule 94(1)(b) of the Patents Rules 2007 provides that:

   The comptroller shall not issue an opinion if the question upon which the opinion is sought appears to him to have been sufficiently considered in any relevant proceedings.

7. Relevant proceedings are defined in Rule 92 as proceedings (whether pending or concluded) before the comptroller, the court, or the European Patent Office.

8. The German Federal Patent Court has already issued a preliminary opinion on the patentability of the patent. In this preliminary opinion, the Federal Patent Court addressed three of the five prior publications mentioned in the request, namely MARKEL-GRAY, MAKHOUL 1 and MAKHOUL 2. The Federal Patent Court preliminarily found that the subject-matter of the patent is new and inventive over any of these three references. Although, it can be argued that these proceedings are not

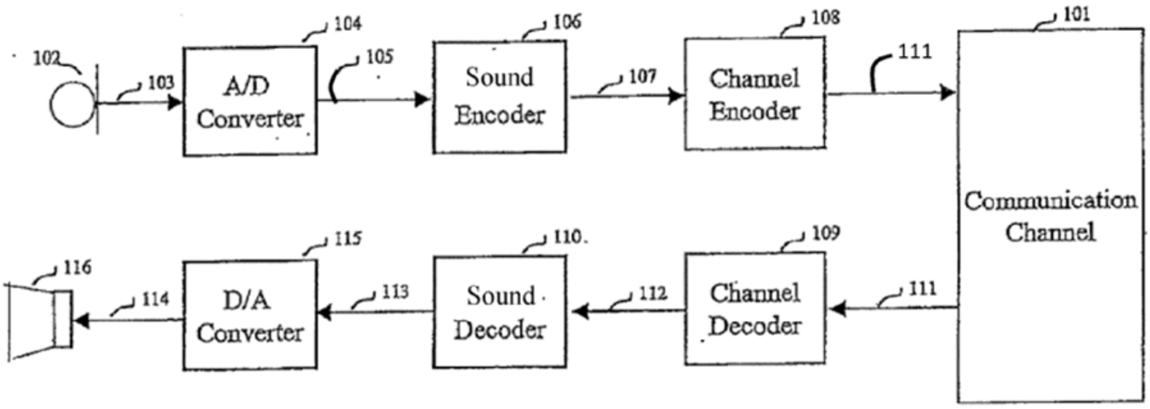
"*relevant* proceedings" under Rule 94(1)(b) as further defined by Rule 92, having reviewed the preliminary opinion of the Federal Patent Court, it appears to me that the question of validity of the patent in the light of these documents has already been sufficiently considered and the preliminary findings of the Federal Patent Court are appropriate in the light of UK Patent Law.

9. Therefore, I have not fully considered again the question of validity of the patent in view of MARKEL-GRAY, MAKHOUL 1 and MAKHOUL 2 inasmuch as this question has already been answered by the Federal Patent Court. Rather, in this opinion, I have summarised the preliminary opinion of the court and outlined my own opinion on the additional arguments of both the requester and observer in view of these documents.

The patent

10. The patent is entitled "Methods, encoder and decoder for linear predictive encoding and decoding of sound signals upon transition between frames having different sampling rates" and was filed on 25 July 2014 with an earlier declared priority date of 17 April 2014. The patent was granted on 26 December 2018 and remains in force in the UK.

11. The patent relates to the field of sound coding. As explained in the patent, and illustrated in the reproduced figure below, a speech encoder 106 converts a speech signal into a digital bit stream that is transmitted over a communication channel 101 (or stored in a storage medium). The speech signal 103 is digitized 104 (sampled and quantized with usually 16-bits per sample) and the speech encoder 106 has the role of representing these digital samples 105 with a smaller number of bits while maintaining a good subjective speech quality. The speech decoder 110 or synthesizer operates on the transmitted or stored bit stream and converts it back to a sound signal 113.



12. One of the best available techniques capable of achieving a good quality/bit rate trade-off is the so-called CELP (Code Excited Linear Prediction) technique (see figure 2 reproduced below). According to this technique, the speech signal 105 is processed in successive frames of samples (corresponding to 10-30 ms of speech), and an LP (Linear Prediction) synthesis filter 216 is computed and transmitted every frame. The frame is further divided into smaller subframes of samples (usually

corresponding to 4-10 ms of speech). An excitation signal 214 is determined in each subframe, which usually comprises two components: one from the past excitation 218 (also called pitch contribution or adaptive codebook) and the other from an innovative codebook 220 (also called fixed codebook). This excitation signal 214 is transmitted and used at the decoder 110 as the input of the LP synthesis filter 216 in order to obtain the synthesized speech 230.

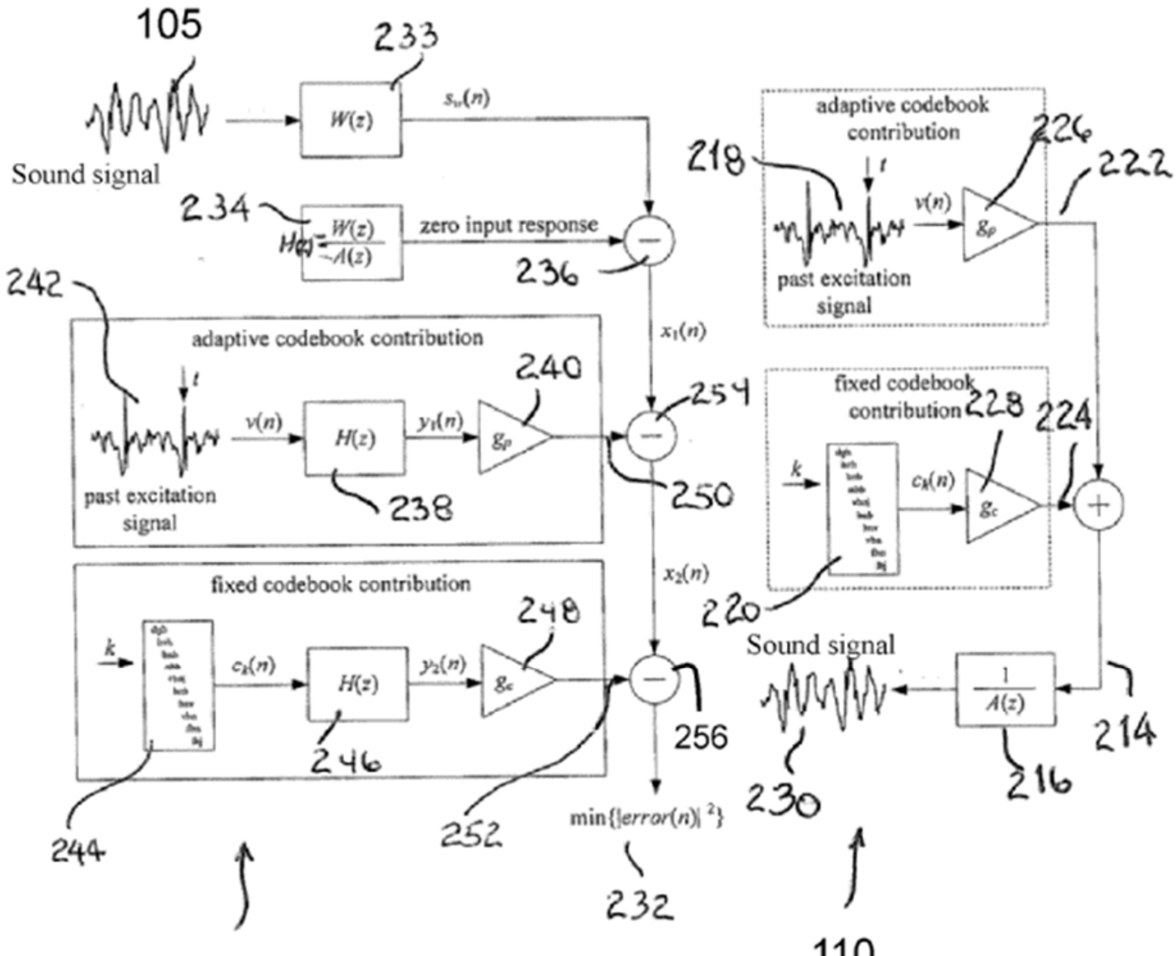


Figure 2

13. In some recent coders, different internal sampling rates are used at different bit rates to improve quality in multi-rate LP-based coding. For example, a multi-rate CELP wideband coder may use 12.8 kHz sampling at bit rates below 16 kbit/s and 16 kHz sampling at bit rates higher than 16 kbits/s. When switching the bit rate between two frames where the internal sampling rate is different, some issues need to be addressed to ensure seamless switching. These issues include the interpolation of LP filter parameters, and the memories of the synthesis filter and the adaptive codebook, which are at different sampling rates. These are issues that the patent seeks to address.

14. The patent includes two independent claims, claims 1 and 10 (with features labelled as in the request), which read:

1. (1) A method implemented in a sound signal encoder or a sound signal *decoder*

(1.1) for converting linear predictive (LP) filter parameters from a sound signal sampling rate S1 to a sound signal sampling rate S2,

the method being characterised by:

(2.1) computing, at the sampling rate S1, a power spectrum of a LP synthesis filter using the LP filter parameters;

(2.2) modifying the power spectrum of the LP synthesis filter to convert it from the sampling rate S1 to the sampling rate S2;

(2.3) inverse transforming the modified power spectrum of the LP synthesis filter to determine autocorrelations of the LP synthesis filter at the sampling rate S2; and

(2.4) using the autocorrelations to compute the LP filter parameters at the sampling rate S2.

10. *(1') A device for use in a sound signal encoder or a sound signal decoder*

(1.1') for converting linear predictive (LP) filter parameters from a sound signal sampling rate S1 to a sound signal sampling rate S2,

the device being characterised in that it comprises:

(2') a processor configured to:

(2.1') compute, at the sampling rate S1, a power spectrum of a LP synthesis filter using the LP filter parameters,

(2.2') modify the power spectrum of the LP synthesis filter to convert it from the sampling rate S1 to the sampling rate S2,

(2.3') inverse transform the modified power spectrum of the LP synthesis filter to determine autocorrelations of the LP synthesis filter at the sampling rate S2, and

(2.4') use the autocorrelations to compute the LP filter parameters at the sampling rate S2.

Claim construction

15. Before I can determine an opinion as to the validity of the patent, I must first construe the claims. This means interpreting the claims in light of the description and drawings as instructed by section 125(1) of the Patents Act:

> *For the purposes of this Act an invention for a patent for which an application has been made or for which a patent has been granted shall, unless the context otherwise requires, be taken to be that specified in a claim of the specification of the application or patent, as the case may be, as interpreted by the description and any drawings contained in that specification, and the extent of the protection conferred by a patent or application for a patent shall be determined accordingly.*

16. I must interpret the claims in context through the eyes of the person skilled in the art. Ultimately, the question is what the person skilled in the art would have understood the patentee to be using the language of the claims to mean. This approach has been confirmed in the recent decisions of the High Court in *Mylan v Yeda*[1] and the Court of Appeal in *Actavis v ICOS*[2].

17. The requester makes reference to a person skilled in the art of speech coding and further defines the skilled person as a scientifically educated engineer. Further, the requester presents document D6 (Vaidyanathan) as demonstrating the general knowledge of the skilled person. The observer provides no formal definition of the skilled person nor their common general knowledge. I note that the preliminary opinion of the German Federal Patent Court considers the person skilled in the art to be an engineer in electrical engineering, communications or information technology (Univ. Master's degree or diploma) with several years of professional experience in the field of digital signal processing, especially for audio coding. I think that this paints a fair picture of the skilled person.

18. *Linear prediction* is a mathematical operation where *predicted* output values are computed from *linear* combinations of past outputs, and present and past inputs. In speech processing, *linear predictive coding (LPC)* is used for representing the spectral envelope of a digital signal of speech in compressed form, using the information of a linear predictive model. In CELP codecs, *A(z)* is commonly referred to as the "*linear predictive (LP) filter*" and the reciprocal 1/*A(z)* is commonly referred to as the "*LP synthesis filter*".

19. Both the method of claim 1 and the device of claim 10 define "*converting linear predictive (LP) filter parameters from a sound signal sampling rate S1 to a sound signal sampling rate S2*" and "*using the LP filter parameters*" to "*compute, at the sampling rate S1, a power spectrum of a LP synthesis filter*". I am of the opinion that a person skilled in the art would understand this to imply that the "*LP filter parameters*" at the "*sound signal sampling rate S1*" are available to either the "*sound signal encoder*" or the "*sound signal decoder*" (in some undefined way) at the start of the conversion process, i.e. before the "*power spectrum of a LP synthesis filter*" is calculated. Paragraph 0025 of the patent indicates that, typically,

1 *Generics UK Ltd (t/a Mylan) v Yeda Research and Dev. Co. Ltd & Anor [2017] EWHC 2629 (Pat)*
2 *Actavis Group & Ors v ICOS & Eli Lilly & Co. [2017] EWCA Civ 1671*

quantized parameters of the LP filter, *A(z)*, are transmitted from the encoder to the decoder and so, although the claim is not limited to such an arrangement, this would be familiar to the skilled person.

Validity – novelty and inventive step

20. Section 1(1) of the Patents Act reads:

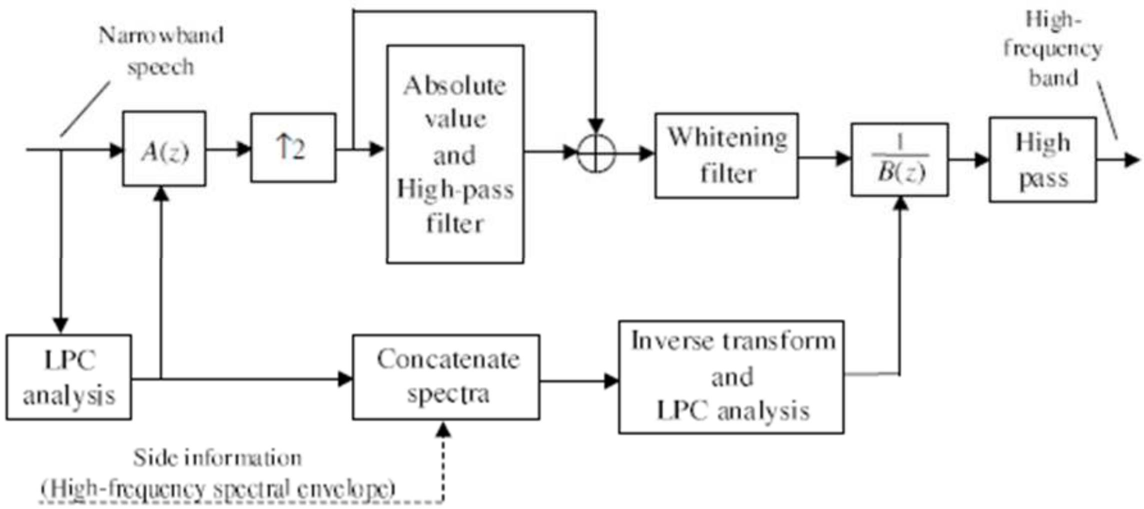
A patent may be granted only for an invention in respect of the following conditions are satisfied, that is to say –

(a) the invention is new;

(b) it involves an inventive step…

21. The requester has argued that claims 1 and 10 of the patent are not novel over VALIN 1, VALIN 2, MARKEL-GRAY, MAKHOUL 1 or MAKHOUL 2. Alternatively, the requester argues that claims 1 and 10 of the patent lack an inventive step over the disclosures of these documents, particularly MAKHOUL 1 and MAKHOUL 2.

22. VALIN 1 presents an algorithm to generate wideband speech (at 16 kHz) from narrowband speech (at 8 kHz) using a small amount of side information. A source excitation model is used for the 3400-8000 Hz band, where the excitation is extrapolated at the receiver and the spectral envelope is transmitted. This is illustrated in the reproduced figure 3 of VALIN 1 below.



23. Section 4.2 of VALIN 1 explains that the power spectrum of an LPC filter is first calculated on the original wideband speech. After proper energy scaling, the 3400-8000 Hz band of this power spectrum is vector quantized in the log-domain using 40 frequency-points (64 points for the whole 0-8 kHz spectrum). The receiver concatenates this high-frequency spectral information with its local calculation of the narrowband spectral envelope. The full-band LPC filter (1/*B*(*z*) in Figure 3) can then be recovered by an inverse transform followed by Levinson-Durbin recursion.

24. The requester also notes that section 1 of VALIN 1 indicates that the limitation of PSTN and wireless networks using the 300-3400 Hz (i.e. narrowband) telephone band is partly due to the 8kHz sampling rate used in those systems, and that an increased audio bandwidth is achievable by increasing the sampling frequency to 16kHz. Additionally, the concluding section 6 of VALIN 1 indicates that the proposed system could be used to transform a narrowband codec into a wideband codec by adding as little as 500 bits/s (i.e. side information) to the transmitted information.

25. The requester argues that a necessary consequence of performing the algorithm proposed in VALIN 1 is that LP filter parameters are converted from a first sampling rate into a second sampling rate. Additionally, according to the requester, the calculated power spectrum of the LP filter is "modified" (i.e. concatenated) at the receiver. The remaining steps of "inverse transforming" and 'determining autocorrelations' are disclosed as outlined above.

26. The observer argues that the bandwidth extension algorithm of VALIN 1 is not implemented in an encoder and that LP parameters are not necessarily transmitted from the transmitter / encoder to the receiver / decoder, such that the narrowband speech upon which LPC analysis is performed at the receiver is not necessarily encoded by linear prediction. However, I note that feature *(1)* of claim 1 (and equivalent feature *(1')* of claim 10) of the patent allows for the claimed invention to be implemented in "*a sound signal encoder <u>or a sound signal decoder</u>*". Furthermore, there is no limitation in claim 1 or claim 10 of the patent that requires LP parameters to be transmitted from a transmitter / encoder to the receiver / decoder (although, as indicated above and referenced at paragraph 0025 of the patent, this would be well-known to the skilled person). Nevertheless, I am of the opinion that reference to "Spectral envelope <u>coding</u>" in section 4.2 of VALIN 1 and "wideband <u>codec</u>" in concluding section 6 of VALIN 1 would be understood by the skilled person to indicate that the algorithm of VALIN 1 could be implemented in either or both an (en)coder and decoder.

27. Furthermore, in my view, section 4.2 of VALIN 1 does refer to steps performed by an encoder. Firstly, this section indicates that "*side information is used to <u>transmit</u> the high-frequency spectral envelope*". Hence, the "*side information*" is transmitted (i.e. by a transmitter / encoder) to the receiver / decoder. Then, this section states that this "*spectral information is <u>transmitted</u>* [i.e. again, by the transmitter / encoder] *in the transform domain. <u>Specifically</u>* [i.e. the 'specifics' of how the "*spectral information is transmitted…*" by the transmitter / encoder, as outlined in the previous sentence]*, the power spectrum of an LPC filter is first calculated on the original wideband speech* [N.B. in my opinion, a skilled person would understand that this is performed by the transmitter / encoder]*. After proper energy scaling, the 3400 – 8000 Hz band of this power spectrum is vector quantized in the log-domain using 40 frequency-points (64 points for the whole 0 – 8 kHz spectrum)* [again, in my view, this would be understood as being performed at the transmitter / encoder]*. The <u>receiver</u>* [i.e. the first action to be performed by the receiver using the transmitted "*side information*"; see figure above] *concatenates <u>this</u> high-frequency spectral information* [i.e. the "*power spectrum*" that was "*calculated*" and "*vector quantized*" at the transmitter / encoder] *with its local calculation of the narrowband spectral envelope…*"

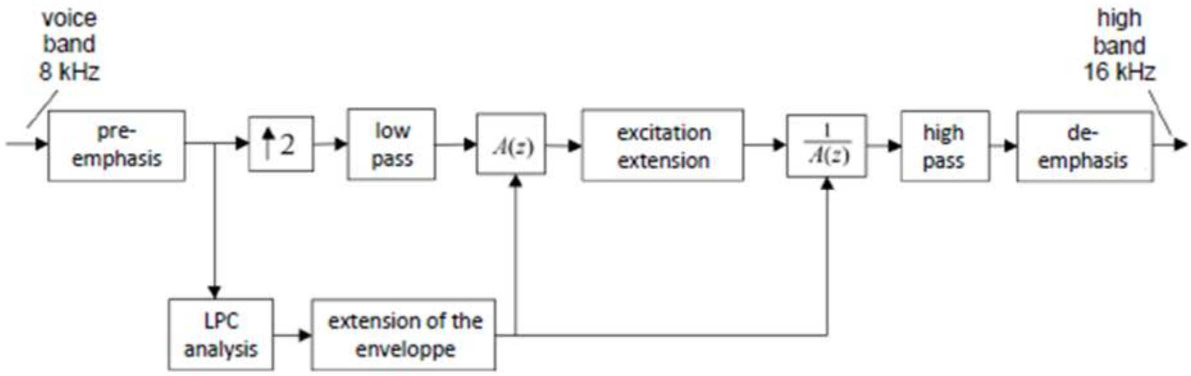
28. However, there is no further detail in VALIN 1 about the calculation of the power spectrum of the LPC filter and I have not been fully convinced by the arguments of

the requester. Particularly, I am not convinced that "*the power spectrum of an LPC filter is first calculated on the original wideband speech*" as disclosed in VALIN 1 is directly equivalent to "*computing, <u>at the sampling rate S1</u>, a power spectrum of a LP synthesis filter using the LP filter parameters*" as required in feature *(2.1)* of claim 1 (and feature (*2.1'*) of claim 10). I notice that the next step of vector quantization is described in a little more detail in the next paragraph of VALIN 1, which reads, "*It was found that an 8-bit vector quantizer was sufficient to transmit a good estimate of the high band… With a frame size of 256 samples at <u>a sampling rate of 16 kHz</u>, the bit rate necessary to transmit this side information is minimal— 500 bits/s.*" But, the necessary detail regarding the sampling rate at which the power spectrum of the LPC filter is calculated is not disclosed.

29. It follows that, in my opinion, '*concatenating*' the *"high-frequency spectral information with its local calculation of the narrowband spectral envelope…*" does not result in "*modifying the power spectrum of the LP synthesis filter to <u>convert it from the sampling rate S1 to the sampling rate S2</u>*". It seems that this step of '*concatenating*' is the linking together of the "*narrowband spectral envelope*", i.e. for the 300-3400 Hz band calculated locally at the receiver, with the "*high-frequency spectral information*", i.e. for the 3400-8000 Hz band received from the transmitter, rather than a '*modification*' resulting in conversion of the sampling rate of the received "*high-frequency spectral information*". N.B. in the patent, this modification is either adding samples to the "*power spectrum*" to increase the sampling rate or removing samples to decrease the sampling rate but, in VALIN, the '*concatenation*' links together the power spectra for 300-3400 Hz and the power spectra for 3400-8000 Hz rather than adding / subtracting samples to / from the 3400-8000 Hz power spectrum.

30. Therefore, based on the arguments presented by the requester, I am not convinced that features *2.1* and *2.2* of claim 1 (and equivalent features *2.1'* and *2.2'* of claim 10) are disclosed in VALIN 1.

31. VALIN 2 is an English translation of a master's thesis for the degree of Master of Applied Science publicly available through the Canadian National Library and Archives. Its content is similar to VALIN 1 but provides some further detail and more background information. For example, section 1.1 of VALIN 2 indicates that, while a sampling frequency of 8kHz is sufficient to transmit a "voice band" signal (i.e. 200Hz to 3500Hz), the AM band (i.e. "wideband"; 50Hz to 7000Hz) requires a sampling frequency of 16kHz. And, the "filter-excitation" model of speech that is used for spectral extension in the high band is shown in the figure reproduced below.



32. Section 2.2.3 of VALIN 2 discloses calculation of the spectral envelope of the signal from linear prediction coefficients, a_i, which corresponds to calculating the spectrum of the impulse response of the synthesis filter 1/*A(z)*. This is done by calculating the spectrum of the response of the analysis filter *A(z)* and then inverting this spectrum. The spectrum of the analysis filter is

$$S_a(k) = \left| \sum_{i=0}^{N} a_i e^{-2\pi j i k/N} \right|^2$$

The power spectrum of the synthesis filter is therefore

$$S_s(k) = \frac{1}{S_a(k)}$$

33. Section 2.2.3 continues to explain that an inverse operation is performed to find the prediction coefficients, a_i, from the power spectrum of the synthesis filter. The autocorrelations can be found by inverse Fourier transform and, once the autocorrelations are found, the prediction coefficients can be retrieved by the Levinson-Durbin recursion.

34. Section 4.3 of VALIN 2 discloses that the high-frequency extension of the spectral envelope consists in estimating a wideband LPC filter (*Fe* = 16 kHz) from a voice band LPC filter (*Fe* = 8 kHz) and some additional parameters, also calculated on the voice band. Section 4.3.2 further outlines that all envelope extension computations are performed in the frequency domain rather than in the domain of the prediction coefficients, a_i, and the transformation shown above is used to represent the synthesis filter, 1/*A(z)*. During processing, the entire 0 - 8000 Hz band is divided into a 64-point spectrum, i.e. at intervals of 125 Hz, with the high band (i.e. 3000-8000 Hz) predicted from the voice band signal (i.e. 250-3500 Hz).

35. However, as for VALIN 1, I do not believe that features *2.1* and *2.2* of claim 1 (and equivalent features *2.1'* and *2.2'* of claim 10) are disclosed in VALIN 2. Crucially, there is no additional detail regarding the calculation of power spectrum of the synthesis filter in Section 2.2.3 so that it does not disclose "*computing, <u>at the sampling rate S1</u>, a power spectrum of a LP synthesis filter using the LP filter parameters*". Furthermore, Section 4.3 does not disclose "*modifying the power spectrum of the LP synthesis filter to <u>convert it from the sampling rate S1 to the sampling rate S2</u>*". The arguments presented by the requester, have not convinced me that these features are present in VALIN 2.

36. MARKEL-GRAY, MAKHOUL 1 and MAKHOUL 2 have all been considered by the German Federal Patent Court, and so I do not intend to reconsider these documents in full. However, having read through both the arguments of the requester and observer, together with the preliminary opinion of the Federal Patent Court, I will summarise my opinion relating to these three documents together.

37. Each of MARKEL-GRAY, MAKHOUL 1 and MAKHOUL 2 provides an explanation of *linear prediction* for speech signals. MARKEL-GRAY is a textbook published in 1976 and, in its most relevant section 6.4 regarding Selective Linear Prediction, it makes repeated reference to MAKHOUL 2. In turn, in relation to claims 1 and 10 of the

patent, I do not believe that MAKHOUL 2 provides anything more significant than the mathematical analysis of speech signals dealt with in MAKHOUL 1.

38. In each case, the German Federal Patent Court came to the preliminary view that the conversion of linear predictive filter parameters from a first sampling rate to a second sampling rate is not disclosed. In my opinion, this view is correct. Although each document does disclose obtaining autocorrelations of a spectral range by inverse transforming a power spectrum of a signal and then using the autocorrelations to calculate prediction coefficients, these steps are not preceded by the calculation of the power spectrum of an LP synthesis filter using LP filter parameters and a calculated power spectrum of the LP synthesis filter is not modified.

39. The requester has argued that, in MARKEL-GRAY, although the algorithm starts from the power spectrum that is calculated from the data sequence spectrum (i.e. representing the input signal), rather than the power spectrum of an LP synthesis filter, elsewhere in the document the skilled person is taught that the LP synthesis filter may alternatively be used instead of the input data. VAIDYANATHAN is cited by the requester (as general knowledge of the skilled person) to indicate that an approximation of power spectrum of the LP synthesis filter is sufficiently similar to the power spectrum of the input signal even at low orders of approximation such that a skilled person would understand that using the LP synthesis filter instead of the input signal would be easier to implement.

40. However, I do not believe that a skilled person would arrive at this conclusion without exercising inventive skill. Contrary to the requester's submission, there is nothing disclosed in MARKEL-GRAY, nor in their common general knowledge, that would lead a skilled person to feature (*2.1*) / (*2.1'*) of claims 1 and 10.

41. The requester further argues that claims 1 and 10 of the patent lack an inventive step over MAKHOUL 1 and MAKHOUL 2.

42. To determine whether or not an invention defined in a particular claim is inventive over the prior art, I will rely on the principles established in Pozzoli SPA v BDMO SA [2007] EWCA Civ 588, in which the well known Windsurfing steps were reformulated:

> *(1)(a) Identify the notional "person skilled in the art";*
>
> *(1)(b) Identify the relevant common general knowledge of that person;*
>
> *(2) Identify the inventive concept of the claim in question or if that cannot readily be done, construe it;*
>
> *(3) Identify what, if any, differences exist between the matter cited as forming part of the "state of the art" and the inventive concept of the claim or the claim as construed;*
>
> *(4) Viewed without any knowledge of the alleged invention as claimed, determine whether those differences constitute steps which would have been obvious to the person skilled in the art.*

43. Steps *(1)(a)* and *(1)(b)* have already been done in paragraphs 17 to 19 above.

44. Step *(2)*: The inventive concept of claims 1 and 10 is converting linear predictive (LP) filter parameters from a sound signal sampling rate S1 to a sound signal sampling rate S2 by modifying a power spectrum of the LP synthesis filter, computed at sampling rate S1, to convert it to sampling rate S2, inverse transforming the modified power spectrum and then using the determined autocorrelations to compute the LP filter parameters at sampling rate S2.

45. Step *(3):* As discussed in paragraph 37 above, neither MAKHOUL 1 nor MAKHOUL 2 disclose modifying a calculated power spectrum of an LP synthesis filter to convert it from a first to a second sampling rate.

46. Step *(4)*: Whilst computing an LP synthesis filter using LP filter parameters would be part of the skilled person's common general knowledge, I believe that the modification of the power spectrum of the LP synthesis filter as claimed in claims 1 and 10 would not have been obvious to the person skilled in the art. As far as I am aware, there is nothing in MAKHOUL 1 or MAKHOUL 2 that would suggest to the skilled person that a conversion of sampling rate could be achieved by modification of the calculated power spectrum.

Opinion

47. In my opinion, based on the arguments presented by the requester, European Patent (UK) 3132443 B1 is valid as claims 1 and 10 are both new and inventive in the light of each of the prior publications outlined in the request.

Dan Hickery
Examiner

NOTE

This opinion is not based on the outcome of fully litigated proceedings. Rather, it is based on whatever material the persons requesting the opinion and filing observations have chosen to put before the Office.