08/540818

Class	Subclass
370	465

ISSUE CLASSIFICATION

[redacted]

5835495

UTILITY SERIAL NUMBER 08/540818

PATENT DATE NOV 10 1998

PATENT NUMBER 5835495

SERIAL NUMBER	FILING DATE	CLASS	SUBCLASS	GROUP ART UNIT	EXAMINER
08/540,818	10/11/95	370	084.0	~~2603~~ ~~2731~~ 2759	Jung

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CONTINUING DATA*******************
VERIFIED None

MJ

FOREIGN/PCT APPLICATIONS**********
VERIFIED None

MJ

CERTIFICATE
APR 18 2000
OF CORRECTION

FOREIGN FILING LICENSE GRANTED 03/28/96

Foreign priority claimed / 35 USC 119 conditions met / Verified and Acknowledged	AS FILED	STATE OR COUNTRY	SHEETS DRWGS.	TOTAL CLAIMS	INDEP. CLAIMS	FILING FEE RECEIVED	ATTORNEY'S DOCKET NO.
☐ yes ☒ no; ☐ yes ☒ no; MJ Examiner's initials	→	WA	6	36	10	$1,778.00	MS1069US

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TITLE

SYSTEM AND METHOD FOR SCALEABLE STREAMED AUDIO TRANSMISSION OVER A NETWORK

U.S. DEPT. OF COMM./PAT. & TM—PTO-436L (Rev.12-94)

PARTS OF APPLICATION FILED SEPARATELY

Applications Examiner: T Coley

NOTICE OF ALLOWANCE MAILED: 6-16-98

Assistant Examiner

CLAIMS ALLOWED

Total Claims	Print Claim
28	7

ISSUE FEE

Amount Due	Date Paid
1320.00	8-31-98

Min Jung
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PATENT EXAMINER
GROUP ~~2600~~ 2731
Primary Examiner

DRAWING

Sheets Drwg.	Figs. Drwg.	Print Fig.
6	6	1

ISSUE BATCH NUMBER #47

Label Area

PREPARED FOR ISSUE

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ISSUE FEE IN FILE

Form PTO-436A
(Rev. 8/92)

(FACE)

RPX Exhibit 1113
RPX v. DAE

08/540818

PATENT APPLICATION

08540818

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INITIALS NOV 30 95

CONTENTS

Date Entered or Counted			Date Received or Mailed
	1.	Application ______ papers.	
	2.	Filing Fee Oath Dec	12-12-95
	3.	~~addt claim~~ NO PAPER	~~12-12-95~~
	4.	Decl Fee Power Atty	1-19-96
	5.	Ext	1-19-96
	6.	IDS w/R	01-19-96
9-18	7.	Rej (3 months) Change of Address	9-4-97 9-22-97
	8.	Amd A	Dec 23 1997 Cert mail 12-17-97
	9.	Final Rej 3 mos	4-14-98 NDH
	10.	Amdt. B (N.E.) (1)	5-15-98 (CM-5-11-98)
6-17=	11.	Ex. Amdt / C	6-17-98 NDH
	12.	PTO GRANT NOV 10 1998	
	13.	Req for C of C	10/21/99
3/7/00	14.	Director's Report	3/21/00
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PATENT APPLICATION SERIAL NO. 08/540818

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE
FEE RECORD SHEET

PTO-1556
(5/87)

BAR CODE LABEL

U.S. PATENT APPLICATION

SERIAL NUMBER	FILING DATE	CLASS	GROUP ART UNIT
08/540,818	10/11/95	370	2603

APPLICANT

PHILIPPE FERRIERE, REDMOND, WA.

CONTINUING DATA*******************
VERIFIED

FOREIGN/PCT APPLICATIONS**********
VERIFIED

FOREIGN FILING LICENSE GRANTED 03/28/96

STATE OR COUNTRY	SHEETS DRAWING	TOTAL CLAIMS	INDEPENDENT CLAIMS	FILING FEE RECEIVED	ATTORNEY DOCKET NO.
WA	6	36	10	$1,778.00	MS1069US

ADDRESS

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TITLE

SYSTEM AND METHOD FOR SCALEABLE STREAMED AUDIO TRANSMISSION OVER A NETWORK

This is to certify that annexed hereto is a true copy from the records of the United States Patent and Trademark Office of the application which is identified above.

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08/540818

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

100

Inventor.. Philippe Ferriere
Applicant.. Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission Over a Network

TRANSMITTAL LETTER AND CERTIFICATE OF MAILING

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
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The following enumerated items accompany this transmittal letter and are being submitted for the matter identified in the above caption.

1. Specification–title page (with listed inventor) plus 51 pages, including 36 claims
2. 6 sheets of formal drawings (Figs. 1-6)
3. Transmittal letter including Certificate of Express Mailing
4. Return Post Card

Large Entity Status [x] Small Entity Status [ ]

Date: Oct. 11, 1995 By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

CERTIFICATE OF MAILING

I hereby certify that the items listed above as enclosed are being deposited with the U.S. Postal Service as either first class mail, or Express Mail if the blank for Express Mail No. is completed below, in an envelope addressed to The Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below-indicated date. Any Express Mail No. has also been marked on the listed items.

Express Mail No. (if applicable) E69853561 91 US

Date: Oct. 11, 1995 By: Lewis C. Lee
Lewis C. Lee

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION FOR LETTERS PATENT

System and Method for Scaleable Streamed Audio Transmission Over a Network

Inventor(s):
Philippe Ferriere

ATTORNEY'S DOCKET NO. MS1-069US

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08/ 518 A/No Fee

TECHNICAL FIELD

This invention relates to audio transmission over networks, such as cable-based and wireless networks used in telephony and computing systems.

BACKGROUND OF THE INVENTION

Digital audio data is transmitted over networks in many different settings. Telephone systems digitize voice and transmit digital voice data over telephone lines or cellular networks. Online service providers on the Internet can download audio files to computer users via conventional telephone or cable lines. Audio files can also be exchanged over traditional data networks, such as LANs (local area networks) and WANs (wide area networks), in a manner akin to electronic mail.

Current implementations of audio file transmission systems involve a transmission scheme in which the audio frames carrying the digital data are a fixed size. Present day modems operate at 9.6 kbps (kilobits per second), 14.4 kbps, and 28.8 kbps. The audio frames from an audio file are compressed at a bit rate for transmission over these various speed communication links. To ensure that transmission is possible over all three conventional speeds, the audio files are typically compressed at a bit rate of 8000 bits/second which can be sent to modems connected at 9.6 kbps, 14.4 kbps, and 28.8 kbps. While this rate will use most of the bandwidth available at 9.6 kbps, it uses only a fraction of the available bandwidth at 14.4 kbps and 28.8 kbps. Since the file is compressed at a lower quality rate of 8000 bits/second, the eventual reconstructed file has an equally low and fixed quality. The customers who use higher performing modems are penalized because they are unable to retrieve audio files of a quality commensurate with the performance of their systems.

It is therefore an aspect of this invention to provide an audio data transmission system which is scaleable to the communication link to use the maximum available bandwidth. In this way, a higher quality audio transmission can be provided to better performing modems.

In the online services setting, conventional systems require transmission of the entire audio file (whether compressed or uncompressed) before the recipient is able to play back the audio file. The audio file is at one fixed quality, such as that provided by the minimal compression rate of 8000 bits/second. For larger audio files carried over limited bandwidth channels (such as low-bandwidth telephone lines), the time required to download the whole audio file can take several minutes. This transmission delay is inconvenient to the recipient, particularly if the recipient is only browsing various audio files with little intent of listening to the entire audio file. The recipient is forced to request an audio file, await the slow transmission of the whole audio file at the minimal fixed bit rate, and then play it back.

Accordingly, it is another aspect of this invention to provide an optimal quality audio streaming in which the recipient can play the audio file as it is received.

SUMMARY OF THE INVENTION

This invention provides an audio file distribution system which permits optimal quality audio file streaming to individual customers with varying modem rates. The audio file distribution system has an audio server which configures the audio files into individual audio data blocks containing a variable number of bits of digital audio data that has been sampled at a selected input sampling rate. The number of bits of digital data and the input sampling rate are scaleable by the audio server to produce an encoded bit stream bit rate that is less than or equal to an effective operational bit rate of a recipient's modem. For example, if the modem connection speed is 14.4 kbps, a version of the audio data compressed at 13000 bits/s might be sent to the recipient; if the modem connection speed is 28.8 kbps, a version of the audio data compressed at 24255 bits/s might be sent to the receiver.

The audio data blocks are then transmitted at the encoded bit stream bit rate to the intended recipient's modem. A computing unit decodes the audio data blocks to reconstruct the audio file and immediately plays the audio file as each audio data block is received. There is no restriction of waiting for the entire audio file to be downloaded before playback. As a result, a customer can request an audio file from the audio server and begin listening immediately. If the customer is just browsing, he/she is free to cancel the audio file before the entire file is transmitted, making the audio file distribution process more efficient and user convenient.

To determine the appropriate block size of the audio data blocks, which enables scaleability to a recipient's effective modem connection speed, the audio server and recipient computing unit are equipped with an audio coder/decoder (or "codec"). The audio codec comprises a coder to encode digital samples representative of an audio input frame into a compressed format for transmission. The coder includes multiple quantizers for encoding the digital samples into the audio data blocks of various sizes, and a quantizer selector to select the appropriate one of the quantizers.

In the illustrated implementation, the coder is configured according to the European Group Speciale Mobile (GSM) standard. This coder has nine quantizers. Each quantizer encodes samples representative of an audio input frame consisting of 160 input audio samples into audio data blocks of a particular size associated with that quantizer. There are nine different block sizes, one for each corresponding quantizer. The block sizes differ according to a number of audio data bits contained in each audio data block. Moreover, each quantizer can be operated to encode the samples for three different input sampling rates. As a result, the coder can output 27 possible encoded bit stream bit rate from the available permutations of nine block sizes and three sampling rates.

The 27 possible encoded bit stream bit rates can be stored in lookup tables at the audio server and recipient computing units. The audio server selects the appropriate combination of block size and input sampling rate from the lookup table which maximizes the bandwidth of the receiving modem. The audio server then uses the selected sampling rate to generate audio samples and chooses the appropriate quantizer to encode those samples into the appropriate block size. The resulting encoded bit stream bit rate provides optimum quality for the receiving modem.

According to another aspect of this invention, a communication system involving multiple communication units and an interconnecting network is also adapted with an audio codec which facilitates scaleable and optimal audio quality real-time communication. An initiating communication unit supplies the effective bit rate of its associated modem to a responding communication unit. The responding communication unit then determines the smallest effective bit rate between the effective bit rates for the modems of the initiating and responding

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communication units, and sends the smallest effective bit rate back to the first communication unit. From that point on, the audio codecs select an appropriate quantizer which produces the audio data blocks with the quantity of bits and input sampling rate that yield an encoded bit stream bit rate of less than or equal to the smallest effective bit rate of the modems. The audio data blocks are then exchanged over the network at the encoded bit stream rate.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a diagrammatic illustration of an audio file distribution system according to one aspect of this invention.

Fig. 2 is a block diagram of an audio coder/decoder (codec) according to another aspect of this invention. The audio codec is illustrated in an example implementation as an RPE-LTP codec according to the European GSM standard.

Fig. 3 is a block diagram of an RPE encoder employed in the Fig. 2 codec.

Fig. 4 is a block diagram of an RPE decoder employed in the Fig. 2 codec.

Fig. 5 is a flow diagram of a method for supplying audio files according to another aspect of this invention.

Fig. 6 is a diagrammatic illustration of a communication system according to another aspect of this invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Fig. 1 shows an audio file distribution system 20 for supplying digital audio files to multiple different participants. Audio file distribution system 20 has a headend 22 with an audio server 24 and an audio file storage 26. System 20 further includes multiple participants 28, 29, and 30 which use services provided by headend 22. Participants 28-30 are each equipped with a corresponding computing unit 32, 33, and 34 and corresponding modem 36, 37, and 38, respectively. The participant computing units are illustrated as desktop computers, but can alternatively be in implemented as other types of personal computers, telephone units, set-top boxes, or other digital processing mechanisms that are capable of handling digital audio data. The participant computing units 32-34 are interconnected to the audio server 24 via a network, represented by network cloud 40. The network 40 might be in the form of a wireless network, such as satellite and cellular phone networks, or a wire-based network, such as low-bandwidth telephone lines or higher-bandwidth cable networks.

As an example, the audio file distribution system 20 might be an online network system in which participants 28-30 dial up and request audio files from the headend 22. The audio server 24 retrieves the audio files from the storage 26 and downloads the audio files to the requesting computing units 32-34. As another example, the audio file distribution system 20 might be implemented as part of an interactive television (ITV) system in which subscribers 28-30 send requests over the TV cable to a cable headend for certain audio files for use in conjunction with, or separate from, video programs.

For discussion purposes, the modems 36-38 each operate at a different modem rate. The three most conventional modem rates are 9.6 kbps (kilobits per second), 14.4 kbps, and 28.8 kbps. Despite these different modem rates, however, the audio file distribution system 20 is capable of supplying the audio files at different bit rates which are appropriate for the receiving modem. More particularly, when requesting an audio file, the requesting computing unit transmits

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its present modem connection speed in terms of effective bit rate, which may be equal to or less than the modem rate. Suppose, for example, that computing unit 33 requests an audio file and its modem 37 is presently operating at an effective bit rate of 13.0 kbps. The computing unit 33 determines this effective bit rate by querying its operating system for the current connection speed of modem 37. The effective bit rate of 13.0 kbps is slightly less than the maximum modem rate of 14.4 kbps. This is not unusual. Often times two modems will negotiate to a slightly lower bit rate, and in cases of modem sharing, modem resources might be partly consumed by other activities thereby explaining a lower effective bit rate.

The computing unit 33 sends a request for an audio file and the effective bit rate of the modem 37 to the audio server 24. In return, the audio server 24 supplies a compressed version of the audio file over network 40 to computing unit 33 such that a bit stream bit rate of the compressed version is less than or equal to the effective bit rate of 13.0 kbps for modem 37. For instance, the audio server 24 might supply the compressed version at a bit rate of 12.955 kbps, 12.1 kbps, or 11.3 kbps.

Now suppose that computing unit 34 sends a request for the same audio file along with an effective bit rate of corresponding modem 38 which is, for example, 27.5 kpbs. In this case, the audio server 24 supplies a compressed version of the audio file over network 40 to computing unit 34 such that a bit stream bit rate of the compressed version is less than or equal to the effective bit rate of 27.5 kbps for modem 37. Here, the audio server 24 might supply the compressed version at a bit rate of 25.9 kbps, 22.6 kbps, or 15.4 kbps.

The audio server 24 thereby provides a compressed version of the requested audio file that is scaled to maximize the available bandwidth of the receiving modem. In the examples, the audio server 24 sent one compressed version of the audio file scaled to the speed of modem 37 (i.e., ≤ 13.0 kbps) and sent a second compressed version of the same audio file scaled to the speed of modem 38 (i.e., ≤ 27.5 kbps). The scaleability permits delivery of variable quality audio files that are commensurate with the communication bandwidth. The audio server 24 provides a higher quality version of the audio file to computing unit 34 (which has a higher performing modem) and a lower quality version to computing unit 33 (which has a lower performing modem).

The compressed audio file supplied from the audio server 24 consists of individual audio data blocks which contain a certain number bits of digital audio data produced at a selected sample rate. The audio data blocks have variable size depending upon the number of data bits included therein. The number of digital audio data bits and the sample rate are selected to provide an encoded bit stream bit rate that is less than or equal to the effective bit rate of the receiving modem.

Upon receipt of the audio data blocks representing the compressed audio file, the computing unit 33 decodes the audio data blocks and reproduces audio sound from the audio data blocks as they are received from the audio server. The computing unit does not wait for the entire file to be downloaded before decompressing the file; but instead plays the audio sound as the blocks are being received. The participant can thus begin listening to a requested audio file immediately, and cancel the file if he/she desires to quit listening to that file and move onto another file. Additionally, by scaleably encoding individual blocks, the receiving computing unit is ensured of optimal quality audio data.

A participant can also request multiple audio files from the audio server 24. In this case, the audio server 24 supplies a compressed version of each audio file

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over network 40 to the requesting computing unit. The bit stream bit rate of the compressed versions of each audio file is less than or equal to the effective bit rate of the receiving modem. Upon receipt of the compressed versions, the computing unit decompresses the audio data blocks, mixes the results, and plays the mixed version.

The audio server 24 and computing units 32-34 are all equipped with an audio coder/decoder (or "codec"). One suitable type of codec is a time-domain codec, and more particularly, an analysis-by-synthesis predictive codec. There are a variety of speech and other audio coding standards for different applications, both nationally and internationally. The standards are based upon different coding rates and employ different types of coders. The audio codecs are configured using one of the common standards, which includes versions of CCITT (International Telephone and Telegraph Consultative Committee), a European GSM (Group Speciale Mobile) standard, CTIA (Cellular Telecommunications Industry Association), and two U.S. federal standards. Table 1 lists various coding standards:

Table 1

Standard	Rate (kbps)	Coder
CCITT G.711	64	log PCM
CCITT G.721	32	ADPCM
CCITT G.723	24, 40	ADPCM
CCITT G.727	16, 24, 32, 40	Embedded ADPCM
CCITT G.728	16	LD-CELP
GSM	13	RPE-LTP
CTIA	8	VSELP
Fed. Std. 1016	4.8	CELP
Fed. Std. 1015	2.4	LPC

The various coders listed in Table 1 are as follows: PCM (pulse code modulation), ADPCM (adaptive differential PCM), LD-CELP (low delay code-excited linear prediction), RPE-LTP (regular pulse excitation - long-term predictor), VSELP (vector sum excited linear prediction), CELP (code-excited linear prediction), and LPC (linear predictive coding). Fig. 2 shows a block diagram of an audio codec 50 which is based in part on the European GSM standard, but modified to perform the encoding/decoding functions required by aspects of this invention. The audio codec 50 is preferably implemented in software which executes on the audio server and recipient computing units. The audio codec 50 encodes input audio frames of 160 audio samples (8-bit or 16-bit PCM format) into audio data blocks of various sizes and decodes the audio data blocks to reconstruct output audio frames of 160 audio samples. In the implementation described herein, the audio data blocks have nine different sizes of

164 bits, 176, bits, 188, bits, 200 bits, 212 bits, 224 bits, 236 bits, 248 bits, and 260 bits. The difference in block sizes are caused by differing numbers of encoded signal sample bits, whereby more bits results in higher quality and fewer bits results in lower quality. However, the omitted bits for the smaller block sizes are selected such that the quality loss is negligible and not too problematic to the human auditory system. The coding scheme is RPE-LTP (regular pulse excitation – long-term predictor).

The audio codec 50 includes an RPE-LTP coder 52 and an RPE-LTP decoder 54. The RPE-LTP coder 52 comprises a preprocessor 56, an LPC (linear predictive coding) analyzer 58, a short term analysis filter 60, a long term predictor filter 62, and an RPE coder 64. The function of all RPE-LTP coder components other than the RPE encoder 64 are standard, and will not be described in detail. Rather, a summary of the functions are provided. A more detailed presentation of these components is described in the ETSI-GSM Technical Specification entitled "GSM Full Rate Speech Transcoding", GSM 06.10, Version 3.2.0, which is hereby incorporated by reference.

Preprocessor 56 receives an input audio frame consisting of 160 signal samples that are sampled at three different input sampling rates of 8000 samples/second, 11025 samples/second, and 22050 samples/second. For the 8000 Hz sampling rate, the 160 input samples represent 20 ms of audio. For the 11025 Hz sampling rate, the 160 input samples represent 14.5 ms of audio. Finally, for the 22050 Hz sampling rate, the 160 input samples represent 7.25 ms of audio. The preprocessor 56 produces an offset-free signal that is then subjected to a first order pre-emphasis filter, such as a FIR (Finite Impulse Response) filter.

The LPC analyzer 58 analyzes the 160 samples to determine coefficients for use in the short term analysis filter 60. The LPC analyzer 58 performs such tasks as segmentation of the audio frame, autocorrelation, calculation of reflection coefficients using the Schur recursion algorithm, transformation of the reflection coefficients into log area ratios LARs, and quantization and coding of the LARs. The short term analysis filter 60 filters the same 160 samples to produce a short term residual signal. The short term analysis filter 60 performs such tasks as decoding the LARs from the LPC analyzer 58, interpolating the decoded LARs to avoid spurious transients which may occur if the filter coefficients are changed abruptly, transforming the LARs into reflection coefficients, and short term analysis filtering.

The audio frame is divided into four sub-frames, with each sub-frame having forty samples of the short term residual signal. The sub-frames are processed blockwise by the long term predictor filter 62 and RPE encoder 64. Each sub-frame is initially passed to the long term predictor (LTP) filter 62. Before processing the sub-frame, LTP parameters used in the LTP filter 62 are estimated and updated using the current sub-frame and a stored sequence of the 120 previous reconstructed short term residual samples. These LTP parameters include LTP lags N and LTP gains b.

A segment of forty long term residual signal samples is obtained by subtracting forty estimates of the short term residual signal from the short term residual signal itself. The resulting segment of forty long term residual samples, designated as "e," is fed to the RPE encoder 64 for compression. The RPE encoder 64 encodes the long term residual samples into a compressed format for transmission. The compressed format contains the RPE parameters which include

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a ~~signal~~ samples x_m, a maximum of the samples x_{max}, and a grid position M, as will be described in more detail below.

The RPE encoder 64 also produces a segment of forty samples of the quantized version of a reconstructed long term residual signal, designated as "e'," and sends the samples back to the LTP filter 62. The forty quantized samples of the long term residual are added to the previous sub-frame of forty short term residual signal estimates to produce a reconstructed version of the current short term residual signal. This sub-frame of reconstructed short term residual signal samples is then fed back to produce a new sub-frame of forty short term residual signal estimates, thereby completing a feedback loop used in predictive coders of this type.

The RPE parameters (x_m, x_{max}, M) and LTP parameters (N, b) for all four sub-frames, along with the filter parameters (LARs), are configured into audio data blocks 66 of various sizes. These audio data blocks are then transmitted to the RPE-LTP decoder 54.

Fig. 3 shows a block diagram of the RPE encoder 64 in more detail. RPE encoder 64 has a weighting filter 70, an RPE grid selector 72, nine quantizers 74_0-74_8, nine corresponding inverse quantizers 76_0-76_8, and an RPE grid positioner 78. The weighting filter 70 is a FIR filter that is applied to each sub-frame by convolving the forty long term residual samples e with an 11-tap impulse response. One suitable impulse response is provided in the above-referenced and incorporated ETSI-GSM Technical Specification. This filtering process yields a filtered signal x.

The RPE grid selector 72 down-samples the filtered signal x by a ratio of three to yield three interleaved sequences consisting of 14, 13, and 13 samples.

The RPE grid selector then splits these sequences into four sub-sequences x_m, where "m" denotes the position of a decimation grid. Each sub-sequence x_m has thirteen RPE samples. The RPE grid selector 72 selects an optimum sub-sequence x_M which has the maximum energy from among the four sub-sequences, where "M" denotes the optimum grid position.

One of the quantizers 74_0-74_8 encodes the sub-sequence of RPE samples into a compressed format for transmission. More particularly, the selected sub-sequence $x_M(i)$ of thirteen RPE samples is quantized by one of the quantizers 74_0-74_8 using APCM (Adaptive Pulse Code Modulation). To perform the quantization, a maximum x_{max} of the absolute value $|x_M(i)|$ is selected for each sub-sequence of thirteen samples $x_M(i)$. The maximum x_{max} is quantized logarithmically and output as one of the RPE parameters in the audio data block 66. The thirteen RPE samples of the selected sub-sequence $x_M(i)$ are then normalized by a decoded version x'_{max} of the block maximum, as follows:

$$x'(i) = x_M(i)/x'_{max}; i = 0,\ldots,12$$

The normalized samples $x'(i)$ are quantized uniformly with one of the nine quantizers 74_0-74_8. The appropriate quantizer is selected by the RPE grid selector 72 depending upon the best available effective bit rate wMaxBitRate at which the receiving modem is presently operating. In this manner, the RPE grid selector also functions as a quantizer selector. The effective bit rate wMaxBitRate of the receiving mode is known by the RPE encoder prior to the quantization process. In the system of Fig. 1, for example, the participant computing units queried the operating system for the present modem connection speed and forwarded this effective bit rate wMaxBitRate to the audio server.

Depending upon which quantizer is selected, the audio data blocks output by the RPE-LTP coder 52 are different in size. The audio data blocks have nine different sizes of 164 bits, 176, bits, 188, bits, 200 bits, 212 bits, 224 bits, 236 bits, 248 bits, and 260 bits depending upon which quantizer is selected. The various sized audio data blocks differ in the number of bits used to represent the thirteen normalized RPE samples for each of the four sub-frames. The smaller audio data blocks (i.e., 164-bit and 176-bit) contain fewer bits to represent the normalized RPE samples, whereas the larger audio data blocks (i.e., 248-bit and 260-bit) contain more bits to represent the normalized RPE samples. The fewer the bits results in a slightly lower quality signal, but not to an annoying or disruptive level.

When the effective bit rate of the receiving modem is comparatively smaller, representing a lower performing modem, a quantizer that causes output of smaller sized audio data blocks is selected. The lower quality signal is commensurate with the performance of the receiving modem. Conversely, when the effective bit rate of the receiving modem is comparatively higher, representing a better performing modem, a quantizer that causes output of larger sized audio data blocks is selected. The higher quality signal is commensurate with the performance of the better performing receiving modem. In this manner, the multiple quantizers enable the RPE-LTP coder 52 to be scaleable according to the awaiting modem capabilities.

The following discussion provides a specific example implementation of the nine quantizers used in an audio RPE-LTP coder that is implemented according to the European GSM standard. The first quantizer 74_0 is selected when the wMaxBitRate is at a level 0. The first quantizer 74_0 uniformly quantizes the first eight normalized samples x'(i) of the first sub-frame to two bits, the last five normalized samples x'(i) of the first sub-frame to one bit, and the thirteen normalized samples x'(i) of the three last sub-frames to one bit. Upon selection of the first quantizer, the RPE-LTP coder 52 will output a 164-bit audio data block having the bit allocation shown in Table 2.

Table 2

wMaxBitRate == 0			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 LARs	LAR1-LAR8	36
	1 LTP lag	N1	7
	1 LTP gain	b1	2
Sub-frame #1 parameters	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	**21**
	1 LTP lag	N2	7
	1 LTP gain	b2	2
Sub-frame #2 parameters	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**13**
	1 LTP lag	N3	7
	1 LTP gain	b3	2
Sub-frame #3 parameters	1 RPE grid position	M3	2

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	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**13**
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**13**
		Total	***164***

The second quantizer 74_1 is selected when the wMaxBitRate is at a level 1. The second quantizer 74_1 uniformly quantizes the thirteen normalized samples x'(i) of the first sub-frame to two bits, the first seven normalized samples x'(i) of the second sub-frame to two bits, the last six normalized samples x'(i) of the second sub-frame to one bit, and the thirteen normalized samples x'(i) of the two last sub-frames to one bit. Upon selection of the second quantizer, the RPE-LTP coder 52 will output a 176-bit audio data block having the bit allocation shown in Table 3.

T19CX

Table 3

wMaxBitRate == 1			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
Sub-frame #1 parameters	1 LTP lag	N1	7
	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	**26**
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**20**
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**13**
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**13**

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	Total	***176***

The third quantizer 74_2 is selected when the wMaxBitRate is at a level 2. The third quantizer 74_2 uniformly quantizes the thirteen normalized samples x'(i) of the first two sub-frames to two bits, the first six normalized samples x'(i) of the third sub-frame to 2 bits, the last seven normalized samples x'(i) of the third sub-frame to one bit, and the thirteen normalized samples x'(i) of the last sub-frame to one bit. Upon selection of the third quantizer, the RPE-LTP coder 52 will output a 188-bit audio data block having the bit allocation shown in Table 4.

Table 4

wMaxBitRate == 2			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
Sub-frame #1 parameters	1 LTP lag	N1	7
	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	**26**
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**26**

Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**19**
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**13**
		Total	***188***

The fourth quantizer 74_3 is selected when the wMaxBitRate is at a level 3. The fourth quantizer 74_3 uniformly quantizes the thirteen normalized samples x'(i) of the first three sub-frames to two bits, the first five normalized samples x'(i) of the last sub-frame to two bits, and the last eight normalized samples x'(i) of the last sub-frame to one bit. Upon selection of the fourth quantizer, the RPE-LTP coder 52 will output a 200-bit audio data block having the bit allocation shown in Table 5.

Table 5

wMaxBitRate == 3			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36

	1 LTP lag	N1	7
	1 LTP gain	b1	2
Sub-frame #1 parameters	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	**26**
	1 LTP lag	N2	7
	1 LTP gain	b2	2
Sub-frame #2 parameters	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**26**
	1 LTP lag	N3	7
	1 LTP gain	b3	2
Sub-frame #3 parameters	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**26**
	1 LTP lag	N4	7
	1 LTP gain	b4	2
Sub-frame #4 parameters	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**18**
		Total	***200***

The fifth quantizer 74_4 is selected when the wMaxBitRate is at a level 4. The fifth quantizer 74_4 uniformly quantizes the first four normalized samples x'(i) of the first sub-frame to three bits, the last nine normalized samples x'(i) of the first sub-frame to two bits, and the thirteen normalized samples x'(i) of the last three sub-frames to two bits. Upon selection of the fifth quantizer, the RPE-LTP coder 52 will output a 212-bit audio data block having the bit allocation shown in Table 6.

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Table 6

wMaxBitRate == 4			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
Sub-frame #1 parameters	1 LTP lag	N1	7
	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	**30**
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**26**
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6

	13 RPE samples	x3(0)-x3(12)	**26**
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**26**
		Total	***212***

The sixth quantizer 745 is selected when the wMaxBitRate is at a level 5. The sixth quantizer 745 uniformly quantizes the thirteen normalized samples x'(i) of the first sub-frame to three bits, the first three normalized samples x'(i) of the second sub-frame to three bits, the last ten normalized samples x'(i) of the second sub-frame to two bits, and the thirteen normalized samples x'(i) of the last two frames to two bits. Upon selection of the sixth quantizer, the RPE-LTP coder 52 will output a 224-bit audio data block having the bit allocation shown in Table 7.

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Table 7

wMaxBitRate == 5			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
Sub-frame #1 parameters	1 LTP lag	N1	7
	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6

	13 RPE samples	x1(0)-x1(12)	39
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	**29**
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**26**
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**26**
		Total	***224***

The seventh quantizer 74_6 is selected when the wMaxBitRate is at a level 6. The seventh quantizer 74_6 uniformly quantizes the thirteen normalized samples x'(i) of the first two sub-frames to three bits, the first two normalized samples x'(i) of the third sub-frame to three bits, the last eleven normalized samples x'(i) of the third sub-frame to two bits, and the thirteen normalized samples x'(i) of the fourth sub-frame to two bits. Upon selection of the seventh quantizer, the RPE-LTP coder 52 will output a 236-bit audio data block having the bit allocation shown in Table 8.

Table 8

wMaxBitRate == 6			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
Sub-frame #1 parameters	1 LTP lag	N1	7
	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	39
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	39
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	**28**
	1 LTP lag	N4	7
	1 LTP gain	b4	2

Sub-frame #4 parameters	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**26**
		Total	***236***

The eighth quantizer 74_7 is selected when the wMaxBitRate is at a level 7. The eighth quantizer 74_7 quantizes the thirteen normalized samples x'(i) of the first three sub-frames to three bits, the first normalized sample x'(i) of the fourth sub-frame to three bits, the last twelve normalized samples x'(i) of the fourth sub-frame to two bits. Upon selection of the eighth quantizer, the RPE-LTP coder 52 will output a 248-bit audio data block having the bit allocation shown in Table 9.

Table 9

wMaxBitRate == 7			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
	1 LTP lag	N1	7
	1 LTP gain	b1	2
Sub-frame #1 parameters	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	39
	1 LTP lag	N2	7
	1 LTP gain	b2	2
Sub-frame #2 parameters	1 RPE grid position	M2	2

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	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	39
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	39
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	**27**
		Total	***248***

The ninth quantizer 74_8 is selected when the wMaxBitRate is at a level 8. The ninth quantizer 74_8 quantizes the thirteen normalized samples x'(i) to three bits. Upon selection of the ninth quantizer, the RPE-LTP coder 52 will output a 260-bit audio data block having the bit allocation shown in Table 10.

Table 10

wMaxBitRate == 8			
Parameter	Parameter name	Variable Name	Number of bits
Filter parameters	8 Log. Area ratios	LAR1-LAR8	36
	1 LTP lag	N1	7

Sub-frame #1 parameters	1 LTP gain	b1	2
	1 RPE grid position	M1	2
	1 Block amplitude	x_{max1}	6
	13 RPE samples	x1(0)-x1(12)	39
Sub-frame #2 parameters	1 LTP lag	N2	7
	1 LTP gain	b2	2
	1 RPE grid position	M2	2
	1 Block amplitude	x_{max2}	6
	13 RPE samples	x2(0)-x2(12)	39
Sub-frame #3 parameters	1 LTP lag	N3	7
	1 LTP gain	b3	2
	1 RPE grid position	M3	2
	1 Block amplitude	x_{max3}	6
	13 RPE samples	x3(0)-x3(12)	39
Sub-frame #4 parameters	1 LTP lag	N4	7
	1 LTP gain	b4	2
	1 RPE grid position	M4	2
	1 Block amplitude	x_{max4}	6
	13 RPE samples	x4(0)-x4(12)	39
		Total	*260*

The nine quantizers 74_0-74_8 can be operated to encode the RPE samples for multiple different input sampling rates. As explained above, there are three input sampling rates in this example implementation: 8000 samples/second, 11025 samples/second, and 22050 samples/second. The appropriate input sampling rate is also determined according to the effective bit rate of the receiving modem. When the effective bit rate of the receiving modem is comparatively smaller, representing a lower performing modem, the selected one of nine quantizers is operated at the lower sample rate to produce a lower quality signal that is commensurate with the performance of the receiving modem. Conversely, when the effective bit rate of the receiving modem is comparatively higher, representing a better performing modem, the selected quantizer is operated at the higher sample rate to produce a higher quality signal that is commensurate with the performance of the receiving modem.

The RPE-LTP coder is therefore scaleable to output the encoded data stream at different bit rates depending upon the effective bit rate of the receiving modem. In this example, there are twenty-seven combinations of block size and input sampling rate. The different bit rates are determined from any combination of the set of nine available block sizes and set of three available input sampling rates. More particularly, the encoded bit stream bit rate EBSBR is determined as follows:

EBSBR = (Block Size x Input Sampling Rate)/Samples in Input Frame

In this example, the input audio data was configured in frames of 160 samples. Based on the 160 sample input frames, the twenty-seven possible bit rates for the corresponding combinations of block size and input sampling rate are provided in Table 11.

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Table 11

Encoded Block Size (bits)	Input Sampling Rate (samples/s)	Encoded Bit Stream Bit Rate (bits/s)
164	8000	8200
176	8000	8800
188	8000	9400
200	8000	10000
212	8000	10600
224	8000	11200
236	8000	11800
248	8000	12400
260	8000	13000
164	11025	11301
176	11025	12128
188	11025	12955
200	11025	13782
212	11025	14609
224	11025	15435
236	11025	16262
248	11025	17089
260	11025	17916
164	22050	22602
176	22050	24255
188	22050	25909

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200	22050	27563
212	22050	29217
224	22050	30870
236	22050	32524
248	22050	34178
260	22050	35832

This table 11 can be stored in the audio server and computing units and utilized as a lookup table to select the desired block size and input sampling rate to produce the encoded bit stream bit rate that will optimize to the receiving modem.

As a continuing example, suppose the audio server 24 (Fig. 1) is configured to encode the audio file in real-time as it is transmitted to a recipient computing unit. For this example, the audio server 24 is loaded with codec software and the above lookup table so that it is suitably programmed to perform the steps shown in Fig. 5. For a given effective bit rate of the receiving modem, the audio server 24 will produce the appropriate sized block and use an appropriate input sampling rate to yield an encoded bit stream bit rate that less than or equal to the receiving modem's effective bit rate, yet greater than the lowest bit rate handled by the RPE encoder 64 (in this case, 8200 samples/second). Recall from the above examples with reference to Fig. 1, the effective bit rates for the receiving modems 37 and 38 were 13.0 kbps and 27.5 kbps, respectively.

The effective bit rates are communicated to the audio server 24 over network 40 (step 150 in Fig. 5). As above, these effective bit rates can be ascertained by the computing units themselves and sent along with a request for an audio file to the headend. The audio server 24 compares the effective bit rate to the encoded bit stream bit rate in the size/rate lookup table 11 (step 152 in Fig. 5). For the 13.0 kbps case, the audio server 24 indexes to an entry where the combination of block size and input sampling rate yields a bit stream bit rate of less than 13000 bits/second. From Table 11, one suitable solution would be a block size of 188 bits and a sample rate of 11025 samples/second which produce an encoded bit stream bit rate of 12955 bits/second.

At step 154, the audio server 24 samples the audio file from the audio file storage 26 at the sampling rate of 11025 Hz to produce audio samples (step 154). The codec 50 executing on the audio server 24 then selects a quantizer to encode these audio samples, whereby the selected quantizer produces the number of bits suitable to generate 188-bit audio data blocks (step 156 in Fig. 5). To achieve this task, the RPE coder employs the third quantizer to encode the correct number of sample bits for the 188-bit blocks (step 158) . The encoded bit stream is then transmitted from the audio sever 24 to the recipient computing unit 33 (step 160).

As an alternative, any one of the combinations listed above this combination in Table 11 can be used. For instance, a combination of 248 bit blocks containing data sampled at 8000 samples/second could be used to provide an encoded bit stream bit rate of 12400 bits/second. For this alternative combination, the audio server samples the audio file at a sampling rate of 8000 Hz and the RPE coder employs the eighth quantizer to encode the audio samples into a correct number of sample bits for the 248-bit audio data blocks.

In the 27.5 kbps case, the audio server 24 selects from the lookup table any combination of block size and input sampling rate that yields a bit stream bit rate of less than 27500 bits/second. From Table 11, a combination of a block size of

188 bits and a sample rate of 22050 samples/second produces an encoded bit stream bit rate of 25909 bits/second. Alternately, combinations above this entry in the table can be used.

With reference again to Fig. 3, the RPE coder 52 further has nine inverse quantizers 76_0-76_8 which perform inverse quantization using APCM. The inverse quantizers decode the quantized bits and denormalize the resulting string using the decoded maximum value x'_{max} leading to the decoded sub-sequence $x'_M(i)$. The RPE grid positioner 78 then up-samples the decoded sub-sequence $x'_M(i)$ by a ratio of three by inserting zero values according to the optimum grid position M to reconstruct forty samples of the long term residual signal e'. The RPE grid positioner 78 outputs the reconstructed long term residual signal e' back to the LTP filter 62.

With reference again to Fig. 2, the RTP-LTP decoder 54 has an RPE decoder 80, a long term predictor filter 82, a short term synthesis filter 84, and a post-processor 86. The RPE decoder 80 decodes and denormalizes the RPE samples using APCM inverse quantization, and then up-samples the decoded sub-sequence to reconstruct forty samples of the long term residual signal e'. The LTP predictor 82 produces the reconstructed short term residual signal for the short term synthesis filter 84, which then reproduces 160 samples representative of the original input audio frame. The post-processor 86 is a deemphasis filter which outputs the 160 sample frame.

Fig. 4 shows the RPE decoder 80 in more detail. It comprises nine inverse quantizers 90_0-90_8 and an RPE grid positioner 92. The inverse quantizers 90_0-90_8 and RPE grid positioner 92 operate essentially identical to the inverse quantizers 76_0-76_8 and RPE grid positioner 78 employed in the feedback loop portion of the RPE encoder 64 described above with reference to Fig. 3. That is, one of the inverse quantizers 90_0-90_8 decodes the quantized bits and denormalizes the resulting string to produce the decoded sub-sequence $x'_M(i)$. The RPE grid positioner 92 then up-samples the decoded sub-sequence $x'_M(i)$ by a ratio of three by inserting zero values according to the optimum grid position M to reconstruct forty samples of the long term residual signal e'.

Fig. 6 shows a communication system 100 that is implemented using a scaleable codec. Communication system 100 has multiple communication units 102, 103, and 104, interconnected by a network 106. The network 106 can be implemented in different ways, including wireless communication networks (such as a cellular network) and wire-based cable networks (such as a telephone network or computer data network). Each communication unit 102-104 is equipped with a corresponding modem 108, 109, and 110 which operate at a modem rate of 28.8 kbps, 28.8 kbps, and 14.4 kbps, respectively.

Suppose an initiating communication unit 102 desires to call a responding communication unit 104. The initiating communication unit 102 queries its operating system for the present effective bit rate of corresponding modem 108 and sends that information to the responding communication unit 104. The responding communication unit 104 then queries its own operating system for the present effective bit rate of corresponding modem 110 and determines the smallest effective bit rate from between the effective bit rates for the modems of the initiating and responding communication units. The responding communication unit 104 returns the smallest effective bit rate back to the initiating communication unit 102. From that point on, real-time communication between the two communication units is established on the best available bit transmission rate that is less than or equal to smallest effective bit rate.

Each communication unit 102-104 is equipped with an I/O soundboard (represented by the telephone handset) that permits uni- or bi-directional audio input/output. Each communication unit is further equipped with a scaleable audio codec 112, 112, and 114, respectively, like the one described above with respect to Figs. 2-4. The audio codecs have multiple quantizers that encode digital data representative of audio sounds into various sized audio data blocks which contain various quantities of bits of sampled audio data. Based upon the discovered least effective bit rate between the communicating pair, the communication units 102 and 104 selects an appropriate sampling rate and data block size that yield the highest quality encoded bit stream bit rate which is still less than or equal to the smallest effective bit rate of the modems. This selection can be facilitated, for example, by the lookup table 11 stored on each communication unit. The soundboards are operated to sample the voice input at the selected sampling rate to produce voice samples. The codecs 112 and 114 then select the quantizer which produces the appropriately sized audio data blocks which, given the input sampling rate, produce the encoded bit stream bit rate which is less than or equal to the smallest effective bit rate of the two modems. This provides the optimal quality for the communication between the two communication units and efficient utilization of the available bandwidth.

It is noted that the computing units of Fig. 1 and the communication units of Fig. 6 can be configured to query the connection speed of their corresponding modems on routine occasions to continually update that information. If the conditions have improved, whereby a modem connection speed increases, the codec might scale the next series of frames to a higher quality which can now be handled by the modems. This dynamic scaling would ensure that the highest quality signal is always being sent.

The scaleable codec described herein with reference to Figs. 2-4 offers many benefits, as demonstrated by the audio file distribution system 20 of Fig. 1 and the communication system 100 of Fig. 6. A first benefit is that the scaleable codec facilitates optimal quality audio file streaming. As described in the Fig. 1 implementation, the audio server encodes audio frames from an audio file stored in a database into audio data blocks of a particular size and an input sampling rate that maximize performance of the recipient's modem. The recipient reconstructs the audio frames from the audio data blocks as the blocks are received, and reproduce the audio sound as the audio data is received. Unlike the prior art systems, there is no limitation that the entire file be transmitted at some low quality fixed bit rate before reproduction and play is possible.

A second benefit is that the scaleable codec facilitates optimal quality audio file streaming and mixing of multiple audio files. A third benefit is that the scaleable codec enables real-time communication, such as videoconferencing, teleconferencing, and computer telephony. This benefit is demonstrated in the Fig. 6 implementation, in which two participants exchange audio frames over a communication link in real-time and play each other's frames as they are received.

In compliance with the statute, the invention has been described in language more or less specific as to structure and method features. It is to be understood, however, that the invention is not limited to the specific features described, since the means herein disclosed comprise exemplary forms of putting the invention into effect. The invention is, therefore, claimed in any of its forms or modifications within the proper scope of the appended claims appropriately interpreted in accordance with the doctrine of equivalents and other applicable judicial doctrines.

CLAIMS

1. A method for encoding digital audio data to be transmitted to a modem operating at an effective bit rate, the method comprising the following steps:

defining an audio data block to carry bits of digital audio data sampled at an input sampling rate;

selecting a block size for the audio data block from among a set of available block sizes so that the block size and input sampling rate determine an encoded bit stream bit rate that is less than or equal to the effective bit rate of the modem; and

encoding the digital audio data into audio data blocks of the selected block size.

2. A method as recited in claim 1 further comprising the following additional steps:

selecting one or more other block sizes from among the set of available block sizes; and

encoding the digital audio data into the audio data blocks of different block sizes.

3. A method as recited in claim 1 wherein the block sizes differ according to a number of bits of digital audio data so that certain audio data blocks contain comparatively more bits of digital audio data than other audio data blocks.

4. A method for encoding digital audio data to be transmitted to a modem operating at an effective bit rate, the method comprising the following steps:

sampling audio data at an input sampling rate to produce audio samples; and

encoding the audio samples into audio data blocks of a block size selected from among a set of available block sizes so that the selected block size and the input sampling rate provide an encoded bit stream bit rate that is less than or equal to the effective bit rate of the modem.

5. A method as recited in claim 4 further comprising the following additional steps:

altering at least one of the block size or the input sampling rate; and

modifying the encoded bit stream bit rate by performing at least one of the following two steps: (1) sampling the audio data at the altered input sampling rate or (2) encoding the audio samples into audio data blocks of a different block size.

6. A method as recited in claim 4 further comprising the step of transmitting the audio data blocks at the encoded bit stream bit rate.

7. A method for supplying digital audio files to a recipient, the recipient having a modem operating to receive digital data at an effective bit rate, the method comprising the following steps:

configuring an audio file into individual audio data blocks, each audio data block containing a certain number bits of digital audio data sampled at an input sampling rate;

selecting a block size for the audio data blocks from among a set of available block sizes and an input sampling rate from among a set of available input sampling rates that determine a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem; and

transmitting the audio data blocks at the bit stream bit rate to the recipient's modem.

~~**8.** A method as recited in claim 7 further comprising the following~~ additional steps:

storing multiple versions of the audio file that are configured in audio data blocks of different block sizes and produced using different input sampling rates; and

choosing an appropriate version that has a block size and an input sampling rate which produces the bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem.

9. A method as recited in claim 7 further comprising the following additional steps:

~~storing the audio file in uncompressed format; and~~

~~configuring an audio file into individual audio data blocks in real-time~~ prior to the transmitting step.

10. A method as recited in claim 7 wherein the transmitting step comprises transmitting the audio data blocks over a distribution network.

11. A method as recited in claim 7 wherein the transmitting step comprises transmitting the audio data blocks over a communication network.

12. A method as recited in claim 7 further comprising the steps of receiving the audio data blocks at the recipient and reproducing audio sound from the audio data blocks as they are received.

13. A method for transmitting multiple digital audio files concurrently to a recipient, the recipient having a modem operating to receive digital data at an effective bit rate, the method comprising the following steps:

performing for each digital audio file the following steps:

configuring the audio file into individual audio data blocks, each audio data block containing a certain number bits of digital audio data sampled at an input sampling rate;

selecting a block size for the audio data blocks from among a set of available block sizes and an input sampling rate from among a set of available input sampling rates that determine a bit stream bit rate;

said ~~selecting step selecting the block~~ size and the input sampling ~~rate for~~ each audio file which ensure that a total bit stream bit ~~rate~~ made up of combined bit stream bit rates of all digital audio ~~files~~ is less than or equal to the effective bit rate of the recipient's modem~~; and~~

transmitt~~ing~~ the audio data blocks for each audio file at the bit stream bit rate ~~to the recipient's modem.~~

12 ~~14~~. A method as recited in claim 11 ~~13~~ further comprising the following additional steps:

receiving the audio data blocks at the recipient; and

reproducing and mixing audio sound from the audio data blocks as they are received.

15. A method for communicating between at least two participants over a network, each participant having a modem operating at an effective bit rate, the method comprising the following steps:

determining a smallest effective bit rate from between the effective bit rates for the modems of the at least two participants;

encoding digital data representative of audio sounds sampled at a selected input sampling rate into audio data blocks of a selected size, the size and the input sampling rate being selected from among respective sets of available sizes and input sampling rates such that an encoded bit stream bit rate of the audio data blocks is less than or equal to the smallest effective bit rate; and

exchanging the audio data blocks between the at least two participants.

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16. An audio file distribution system comprising:

an audio file database to store audio files;

an audio server to retrieve the audio files from the audio file database and supply the audio files in a compressed format;

a client computing unit located remotely from the audio server, the client computing unit having a modem operating to receive digital data at an effective bit rate;

a network interconnecting the audio server and the client computing unit;

said audio server being configured to supply an audio file as individual audio data blocks which contain a certain number bits of digital audio data that have been sampled at a selected sampling rate wherein the number of bits of digital data and the sampling rate are selected to provide an encoded bit stream bit rate that is less than or equal to the effective bit rate of the client's modem; and

said audio server further being configured to transmit the encoded audio data blocks over the distribution network to the client computing unit.

17. An audio file distribution system as recited in claim 16 wherein the client computing unit queries the modem to determine the effective bit rate.

18. An audio file distribution system as recited in claim 16 wherein the client computing unit initially sends a request for a particular audio file along with the effective bit rate of the modem to the audio server.

19. An audio file distribution system as recited in claim 16 wherein:

the audio file database stores multiple versions of each audio file that are configured in audio data blocks of different block sizes and produced using different sampling rates; and

the audio server selects an appropriate version of the audio file that has a block size and sampling rate which produces the bit stream bit rate that is less than or equal to the effective bit rate of the client's modem.

20. An audio file distribution system as recited in claim 16 wherein:

the audio file database stores the audio files in uncompressed format; and

the audio server is configured to encode the audio file into individual audio data blocks in real-time.

21. An audio file distribution system as recited in claim 16 wherein the audio server has a size/rate lookup table listing various combinations of block sizes and sampling rates indexed with associated encoded bit stream bit rates, and the audio server is configured to select the block size and sampling rate using the size/rate lookup table.

22. An audio file distribution system as recited in claim 16 wherein the client computing unit decodes the audio data blocks and reproduces audio sound therefrom as the audio data blocks are received from the audio server.

Sub A3

~~23.~~ An audio ~~file distribution~~ system as recited ~~in claim 16 wherein:~~

the audio server encodes multiple digital audio ~~files~~ into audio data blocks; and

the ~~client~~ computing unit is configured to decode the audio data blocks into audio frames, mix the audio frames, and reproduce ~~sound from the audio frames.~~

20 ~~24~~. A communication system comprising:

first and second communication units, each communication unit being equipped with a modem operating to receive and transmit data at an effective bit rate;

a network interconnecting the first and second communication units;

said first communication unit being configured to supply the effective bit rate of the modem for the first communication unit to the second communication unit;

said second communication unit being configured to determine a smallest effective bit rate from between the effective bit rates for the modems of the first and second communication units and to send the smallest effective bit rate back to the first communication unit;

said first and second communication units being configured to generate digital audio samples representative of an audio signal at a selected input sampling rate, the first and second communication units being equipped with an audio coder/decoder having multiple quantizers that encode the digital audio samples into various sized audio data blocks which contain various quantities of bits of the audio samples, said first and second communication units using an appropriate input sampling rate and selecting an appropriate quantizer to encode the audio samples into audio data blocks that yield an encoded bit stream bit rate less than or equal to the smallest effective bit rate of the modems; and

said first and second communication units exchanging the audio data blocks over the network.

21 ~~25.~~ A communication system as recited in claim 20 ~~24~~ wherein the network is a wireless communication network.

22 ~~26.~~ A communication system as recited in claim 20 ~~24~~ wherein the network is a wire-based data network.

23 ~~27.~~ A communication system as recited in claim 20 ~~24~~ wherein the audio coder/decoder comprises a predictive audio coder/decoder.

24 ~~28.~~ A communication system as recited in claim 20 ~~24~~ wherein the first and second communication units are configured to reproduce the audio signal from the audio data blocks as the audio data blocks are received.

29. A predictive audio coder for encoding digital audio samples representative of an audio signal that has been sampled at an input sampling rate, the predictive audio coder comprising:

a predictor filter to adaptively compute predictive parameters based upon the digital audio samples;

an encoder to encode the digital audio samples into a compressed format for transmission, the encoder comprising:

(a) multiple quantizers for encoding the digital audio samples into various numbers of bits to be contained in audio data blocks so that one quantizer produces a first number of bits for an audio data block of a first size and another quantizer produces a second number of bits for an audio data block of a second size; and

(b) a quantizer selector to select one of the multiple quantizers.

30. A predictive audio coder as recited in claim 29 wherein the predictor filter comprises both long and short term predictive filters.

31. A predictive audio coder as recited in claim 29 configured to conform to the Group Speciale Mobile (GSM) standard, the predictive audio coder comprising nine quantizers.

32. A predictive audio coder as recited in claim 31 wherein:

a first quantizer produces 60 bits for audio data blocks consisting of 164 bits;

a second quantizer produces 72 bits for audio data blocks consisting of 176 bits;

a third quantizer produces 84 bits for audio data blocks consisting of 188 bits;

a fourth quantizer produces 96 bits for audio data blocks consisting of 200 bits;

a fifth quantizer produces 108 bits for audio data blocks consisting of 212 bits;

a sixth quantizer produces 120 bits for audio data blocks consisting of 224 bits;

a seventh quantizer produces 132 bits for audio data blocks consisting of 236 bits;

a eighth quantizer produces 144 bits for audio data blocks consisting of 248 bits; and

a ninth quantizer produces 156 bits for audio data blocks consisting of 260 bits.

33. In an audio coder/decoder implemented using regular pulse excitation (RPE), an RPE encoder comprising:

multiple quantizers for encoding digital audio samples sampled at a selected input sampling rate into a compressed format for transmission, the quantizers encoding the audio digital data into audio data blocks containing a defined number of samples with varying numbers of sample bits so that one quantizer produces an audio data block having a first number of sample bits and another quantizer produces an audio data block having a second number of sample bits different than the first number of sample bits; and

a quantizer selector to select one of the multiple quantizers.

34. An RPE encoder as recited in claim 33 configured to be implemented in an audio coder/decoder conforming to the Group Speciale Mobile (GSM) standard, further comprising nine quantizers.

35. An RPE encoder as recited in claim 34 wherein:

a first quantizer produces audio data blocks consisting of 164 bits including 60 sample bits;

a second quantizer produces audio data blocks consisting of 176 bits including 72 sample bits;

a third quantizer produces audio data blocks consisting of 188 bits including 84 sample bits;

a fourth quantizer produces audio data blocks consisting of 200 bits including 96 sample bits;

a fifth quantizer produces audio data blocks consisting of 212 bits including 108 sample bits;

a sixth quantizer produces audio data blocks consisting of 224 bits including 120 sample bits;

a seventh quantizer produces audio data blocks consisting of 236 bits including 132 sample bits;

a eighth quantizer produces audio data blocks consisting of 248 bits including 144 sample bits; and

a ninth quantizer produces audio data blocks consisting of 260 bits including 156 sample bits.

36. An RPE-LTP coder/decoder incorporating the RPE encoder as recited in claim 33.

add art

08/540818

ABSTRACT

An audio data transmission system encodes audio files into individual audio data blocks which contain a variable number bits of digital audio data that were sampled at a selectable sample rate. The number of bits of digital data and the input sampling rate are scaleable to produce an encoded bit stream bit rate that is less than or equal to an effective operational bit rate of a recipient's modem. The audio data transmission system uses computing units which are designed to select an appropriate combination of block size and input sampling rate to maximize the available bandwidth of the receiving modem. For example, if the modem connection speed for one modem is 14.4 kbps, a version of the audio data compressed at 13000 bits/s might be sent to the recipient; if the modem connection speed for another modem is 28.8 kbps, a version of the audio data compressed at 24255 bits/s might be sent to the receiver. The audio data blocks are then transmitted at the encoded bit stream bit rate to the intended recipient's modem. The audio data blocks are decoded at the recipient to reconstruct the audio file and immediately play the audio file as it is received. The audio data transmission system can be implemented in online service systems, ITV systems, computer data network systems, and communication systems.

PRINT OF DRAWINGS
AS ORIGINALLY FILED

MS1-069US

08/540818

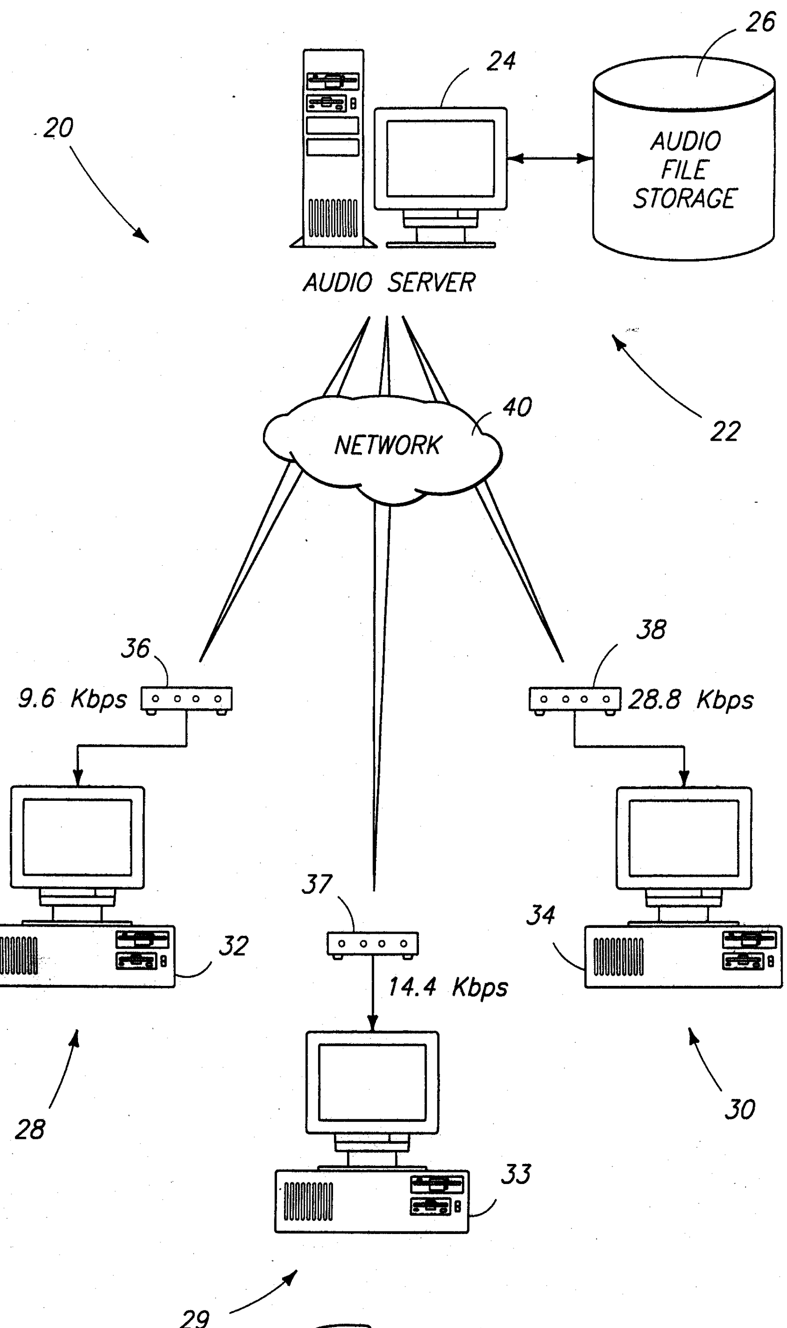

Fig 1

audio file distribution system

5-6
8-14
19-28
37-45

46-48

PRINT OF DRAWINGS
AS ORIGINALLY FILED

08/540818

MS1-069US

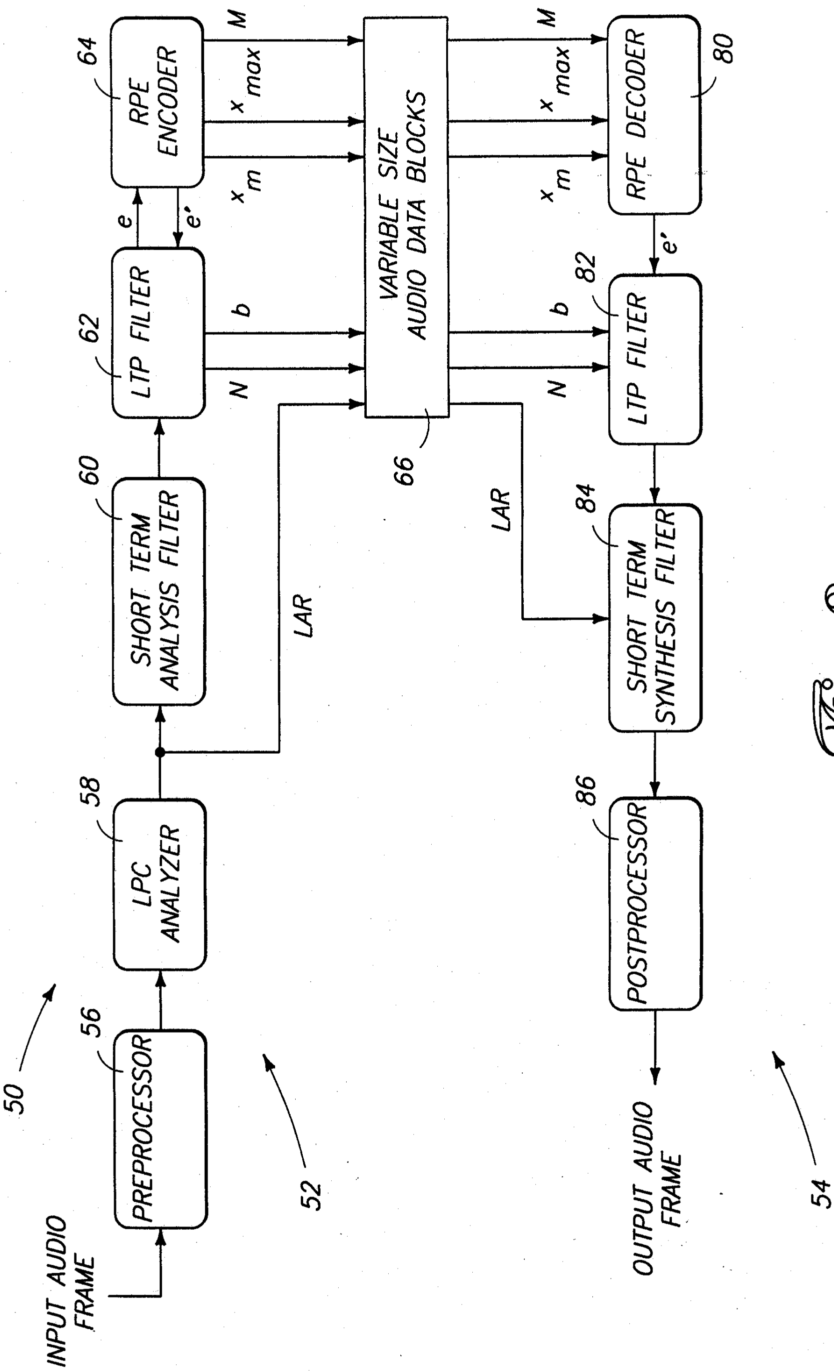

Fig 2

audio codec

PRINT OF DRAWINGS
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MS1-069US

08/540818

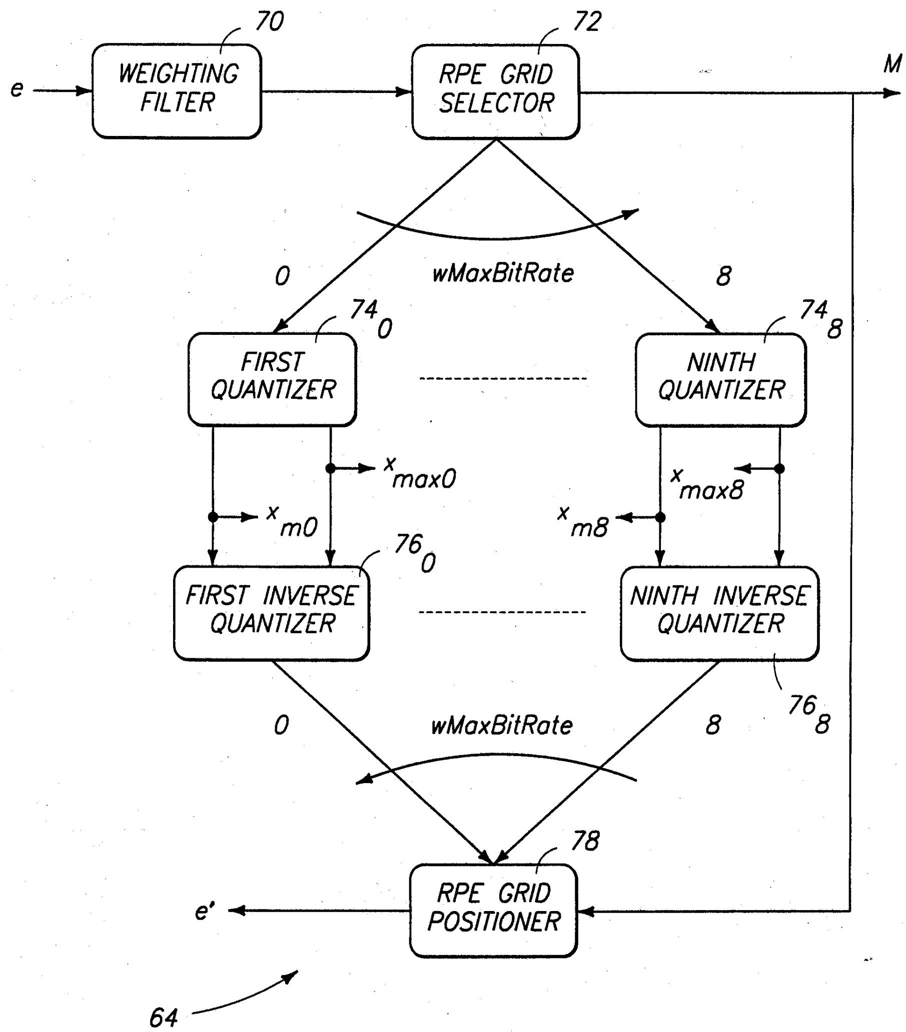

Fig 3

RPE encoder

PRINT OF DRAWINGS
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MS1-069US

08/540818

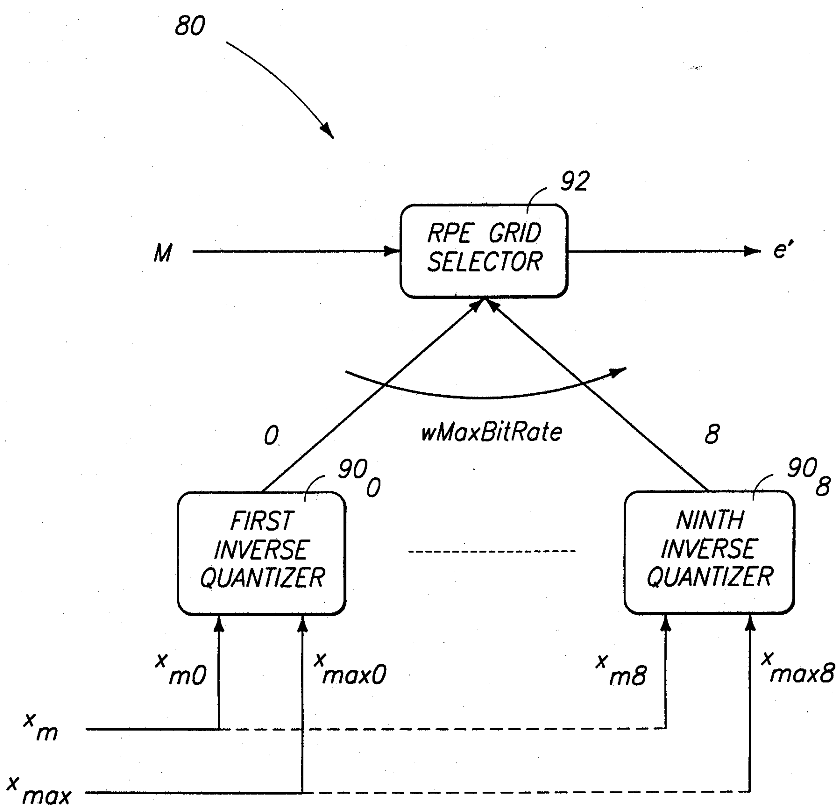

Fig 4

RPE decoder

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08/540818

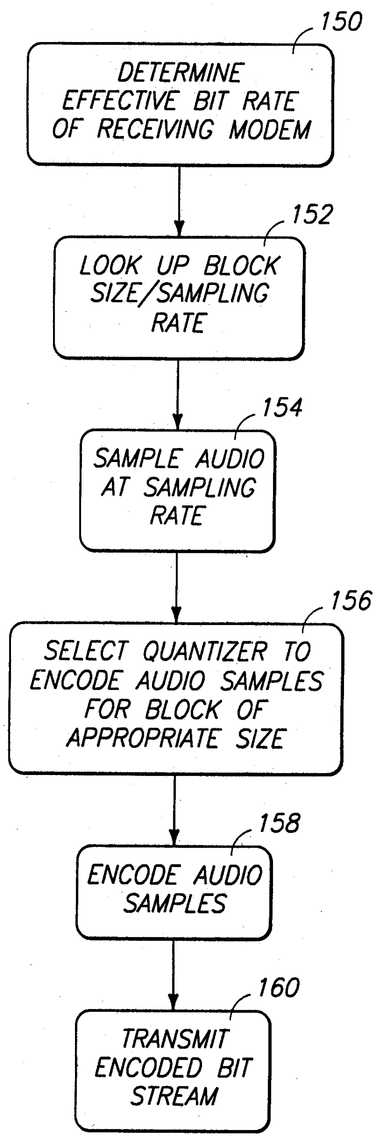

Fig 5

PRINT OF DRAWINGS
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MS1-069US

09/540818

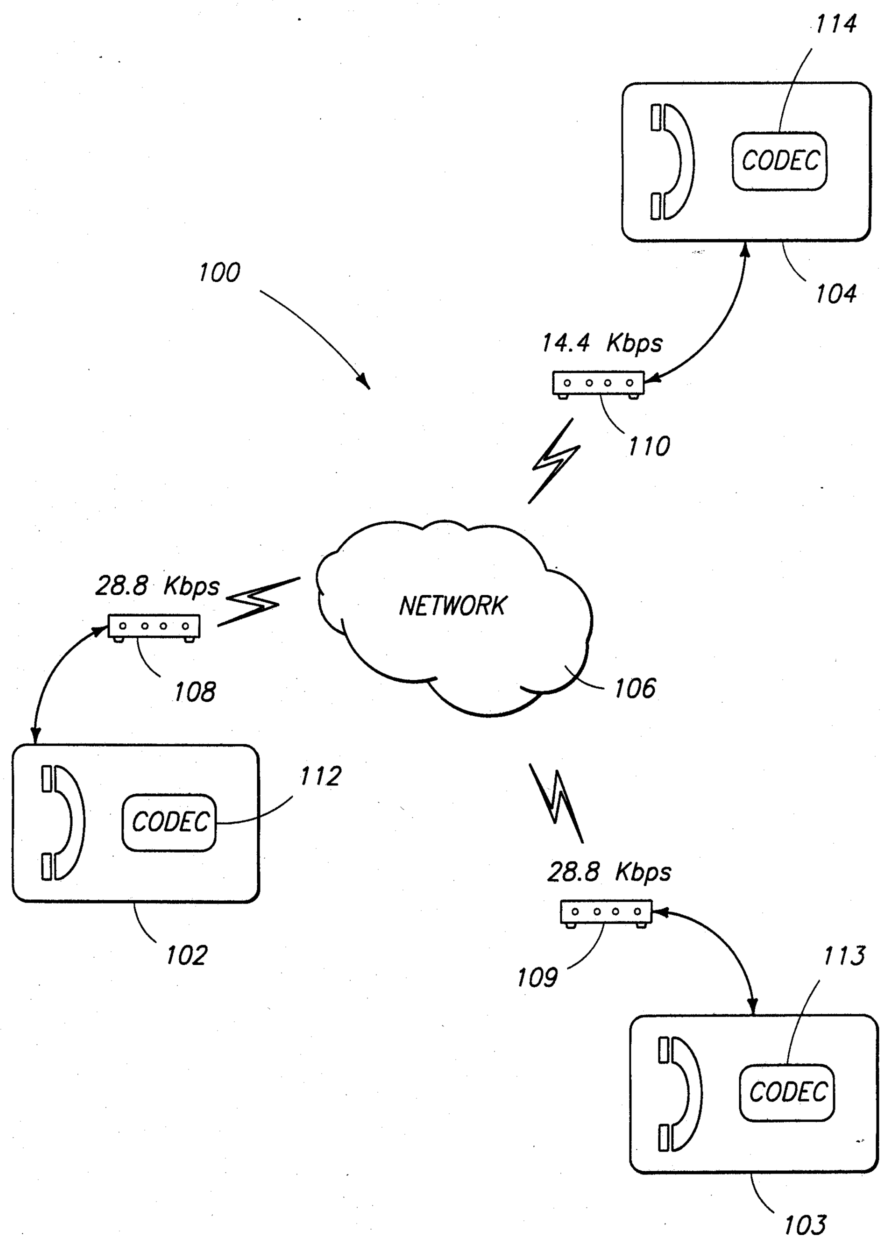


Fig 6

08/540,818 10/11/95

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

#2

APPLICATION NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE

LEE & HAYES
W 1818 FRANCIS #160
SPOKANE WA 99205

0000

12/12/95

DATE MAILED:

NOTICE TO FILE MISSING PARTS OF APPLICATION FILING DATE GRANTED

An Application Number and Filing Date have been assigned to this application. However, the items indicated below are missing. The required items and fees identified below must be timely submitted **ALONG WITH THE PAYMENT OF A SURCHARGE** for items 1 and 3-6 only of $ 130 for large entities or $ 65 for small entities who have filed a verified statement claiming such status. The surcharge is set forth in 37 CFR 1.16(e).

If all required items on this form are filed within the period set below, the total amount owed by applicant as a ☑ large entity, ☐ small entity (verified statement filed), is $ 1778.

Applicant is given **ONE MONTH FROM THE DATE OF THIS LETTER, OR TWO MONTHS FROM THE FILING DATE** of this application, **WHICHEVER IS LATER,** within which to file all required items and pay any fees required above to avoid abandonment. Extensions of time may be obtained by filing a petition accompanied by the extension fee under the provisions of 37 CFR 1.136(a).

1. ☑ The statutory basic filing fee is: ☑ missing ☐ insufficient. Applicant as a ☑ large entity ☐ small entity, must submit $ 750 to complete the basic filing fee.
2. ☑ Additional claim fees of $ 898 as a ☐ large entity, ☐ small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due.
3. ☑ The oath or declaration:
   ☑ is missing.
   ☐ does not cover items omitted at time of execution.

   An oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date is required.
4. ☐ The oath or declaration does not identify the application to which it applies. An oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.
5. ☐ The signature to the oath or declaration is: ☐ missing; ☐ a reproduction; ☐ by a person other than the inventor or a person qualified under 37 CFR 1.42, 1.43, or 1.47. A properly signed oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.
6. ☐ The signature of the following joint inventor(s) is missing from the oath or declaration:

   ______________________ An oath or declaration listing the names of all inventors and signed by the omitted inventor(s), identifying this application by the above Application Number and Filing Date, is required.
7. ☐ The application was filed in a language other than English. Applicant must file a verified English translation of the application and a fee of $__________ under 37 CFR 1.17(k), unless this fee has already been paid.
8. ☐ A $__________ processing fee is required for returned checks. (37 CFR 1.21(m)).
9. ☐ Your filing receipt was mailed in error because check was returned without payment.
10. ☐ The application does not comply with the Sequence Rules. See attached Notice to Comply with Sequence Rules 37 CFR 1.821-1.825.
11. ☐ Other.

Direct the response and any questions about this notice to____________________, Application Processing Division, Special Processing and Correspondence Branch (703) 308-1202.

A copy of this notice MUST be returned with the response.

FORM PTO-1533 (REV. 5-93)

OFFICE COPY

THE FOLLOWING WAS MISSING
FROM THE ORIGINAL USPTO
FILE HISTORY
Entry #3 is Crossed
out on the Table

MAIL ROOM JAN 19 1996 32 PAT. & TRADEMARK OFFICE

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

TRANSMITTAL LETTER AND CERTIFICATE OF MAILING

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 1818 Francis #160, Spokane, WA 99205

The following enumerated items accompany this transmittal letter and are being submitted for the matter identified in the above caption.

1. Transmittal letter including Certificate of Mailing
2. Notice to File Missing Parts of Application
3. Petition for Extension of Time
4. Executed Declaration
5. Check in the Amount of $1888 for Filing Fee ($750), Additional Claims ($898), Notice Fee ($130), and Extension Fee ($110)
6. Assignment with Recordation Form Cover Sheet
7. Check in the Amount of $40 for Recordation Fee
8. Power of Attorney, including a Statement of the Right of Assignee to Prosecute
9. Information Disclosure Statement including PTO-1449 form and referenced prior art
10. Return Post Card

Large Entity Status [x] Small Entity Status [ ]

If necessary, the PTO is hereby authorized to charge any fees required or credit any overpayment to Deposit Account 12-0769 pursuant to 37 CFR 1.25.

Date: Jan. 19, 1996 By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

CERTIFICATE OF MAILING

I hereby certify that the items listed above as enclosed are being deposited with the U.S. Postal Service as either first class mail, or Express Mail if the blank for Express Mail No. is completed below, in an envelope addressed to The Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below-indicated date. Any Express Mail No. has also been marked on the listed items.

Express Mail No. (if applicable) EG576129108US

Date: Jan 19, 1996 By: Lewis C. Lee
Lewis C. Lee

EG576129108US

MAIL ROOM 32 JAN 19 1996 PAT. & TRADEMARK OFFICE

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

DECLARATION FOR PATENT APPLICATION

As a below named inventor, I hereby declare that:

My residence, post office address, and citizenship are as stated below next to my name.

I believe I am the original, first, and sole inventor (if only one name is listed below) or an original, first, and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled "System and Method for Scaleable Streamed Audio Transmission over a Network", the application of which is identified above.

I have reviewed and understand the content of the above-identified specification, including the claims.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, § 1.56(a).

PRIOR FOREIGN APPLICATIONS: no applications for foreign patents or inventor's certificates have been filed prior to the date of execution of this declaration.

All statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statement may jeopardize the validity of the application or any patent issued therefrom.

* * * * * * * * * *

Full name of inventor: Philippe Ferriere,

Inventor's Signature: ______________________

Date: 01/11/96

Residence: 4306 156th Ave. N.E. #YY174

Redmond, WA 98052

Citizenship: France

#4

EG 576129108US

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

POWER OF ATTORNEY

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 1818 Francis #160
Spokane, WA 99205

Microsoft Corporation of Redmond, Washington, a corporation organized under the laws of the State of Washington, certifies that it is the assignee of the entire right, title, and interest in the patent application identified above, by virtue of an Assignment from inventor(s) Philippe Ferriere. A copy of the Assignment is attached.

The undersigned has reviewed the attached Assignment and, to the best of undersigned's knowledge and belief, title to the U.S. patent application identified above is in the name of Microsoft Corporation.

The undersigned is empowered to act on behalf of Microsoft Corporation.

As assignee and owner of all rights to the above-identified application, Microsoft Corporation hereby appoints the following attorneys to prosecute this application and transact all future business in the Patent and Trademark Office related to this application: Lewis C. Lee, Reg. No. 34,656, and Daniel L. Hayes, Reg. No. 34,618.

Send correspondence to: LEE & HAYES, PLLC, W. 1818 Francis #160, Spokane, Washington, 99205. Direct telephone calls to: Lewis C. Lee (509) 324-9256.

I hereby declare that all statements made herein of my own knowledge are true, and that all statements made on information and belief are believed to be true; and further, that these statements are made with the knowledge that willful false statements, and the like so made, are punishable by fine or imprisonment, or both, under section 1001, Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

Microsoft Corporation

Dated: 1/12/96 By: Stephen G. Allen

Stephen G. Allen
Patent Manager

MAIL ROOM
JAN 19 1996
PAT. & TRADEMARK OFF.

$750.00 -101
130.00 -105 EG576129108US

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231
898.00 -103

APPLICATION NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
08/540,818	10/11/95		MSI-069US

LEWIS C LEE
LEE & HAYES
W 1818 FRANCIS #160
SPOKANE WA 99205

0000

12/12/95

DATE MAILED:

NOTICE TO FILE MISSING PARTS OF APPLICATION
FILING DATE GRANTED

An Application Number and Filing Date have been assigned to this application. However, the items indicated below are missing. The required items and fees identified below must be timely submitted **ALONG WITH THE PAYMENT OF A SURCHARGE** for items 1 and 3-6 only of $130 for large entities or $65 for small entities who have filed a verified statement claiming such status. The surcharge is set forth in 37 CFR 1.16(e).

If all required items on this form are filed within the period set below, the total amount owed by applicant as a ☑ large entity, ☐ small entity (verified statement filed), is $1778.

Applicant is given **ONE MONTH FROM THE DATE OF THIS LETTER, OR TWO MONTHS FROM THE FILING DATE** of this application, **WHICHEVER IS LATER,** within which to file all required items and pay any fees required above to avoid abandonment. Extensions of time may be obtained by filing a petition accompanied by the extension fee under the provisions of 37 CFR 1.136(a).

1. ☑ The statutory basic filing fee is: ☑ missing ☐ insufficient. Applicant as a ☑ large entity ☐ small entity, must submit $750 to complete the basic filing fee.

2. ☑ Additional claim fees of $898 as a ☐ large entity, ☐ small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due.

3. ☑ The oath or declaration:
   ☑ is missing.
   ☐ does not cover items omitted at time of execution.

   An oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date is required.

4. ☐ The oath or declaration does not identify the application to which it applies. An oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.

5. ☐ The signature to the oath or declaration is: ☐ missing; ☐ a reproduction; ☐ by a person other than the inventor or a person qualified under 37 CFR 1.42, 1.43, or 1.47. A properly signed oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.

6. ☐ The signature of the following joint inventor(s) is missing from the oath or declaration:

   ________________ An oath or declaration listing the names of all inventors and signed by the omitted inventor(s), identifying this application by the above Application Number and Filing Date, is required.

7. ☐ The application was filed in a language other than English. Applicant must file a verified English translation of the application and a fee of $__________ under 37 CFR 1.17(k), unless this fee has already been paid.

8. ☐ A $__________ processing fee is required for returned checks. (37 CFR 1.21(m)).

9. ☐ Your filing receipt was mailed in error because check was returned without payment.

10. ☐ The application does not comply with the Sequence Rules. See attached Notice to Comply with Sequence Rules 37 CFR 1.821-1.825.

11. ☐ Other.

Direct the response and any questions about this notice to ____________________, Application Processing Division, Special Processing and Correspondence Branch (703) 308-1202.

A copy of this notice MUST be returned with the response.

FORM PTO-1533 (REV. 5-93)

COPY TO BE RETURNED WITH RESPONSE

$/ 10.00 EG571129108US

#5

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

PETITION FOR EXTENSION OF TIME

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 1818 Francis #160
Spokane, WA 99205

Pursuant to 37 C.F.R. 1.136(a), Applicant respectfully petitions for a one month extension of time to respond to the Notice to File Missing Parts which was mailed on December 12, 1995 for the above identified application filed on October 11, 1995. The due date for the Response was originally January 12, 1995 (one month from the date of the Notice), but is respectfully extended an additional one month to February 12, 1996. A payment of $240 for the missing parts fee ($130) and the one-month extension fee ($110) is included in the check for the filing fee.

Respectfully submitted,

Date: Jan. 19, 1996

By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

260 YC 02/01/96 08540818
1 115 110.00 CK

MAIL ROOM 32 JAN 19 1996 PAT. & TRADEMARK OFFICE

#5/IDS 08/4P
Godm 05/29/96

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

INFORMATION DISCLOSURE STATEMENT

References -- See Attached Form PTO-1449

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 1818 Francis #160
Spokane, WA 99205

The attached form PTO-1449 is submitted in compliance with Applicant's duty of disclosure under 37 CFR §1.56.

Dated: Jan. 19, 1996

By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

Sheet 1 of 1

Form PTO-1449	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTY. DOCKET NO. MS1-069US	SERIAL NO. 08/540,818
LIST OF ART CITED BY APPLICANT (Use several sheets if necessary)		APPLICANT Microsoft Corporation	
		FILING DATE October 11, 1995	GROUP 2603

MAIL ROOM 32 JAN 19 1996 PAT. & TRADEMARK OFF.

U.S. PATENT DOCUMENTS

*Examiner Initial		Document Number	Date	Name	Class	Subclass	Filing Date If Appropriate
	AA						
	AB						
	AC						
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						

FOREIGN PATENT DOCUMENTS

		Document Number	Date	Country	Class	Subclass	Translation Yes	Translation No
	AL							
	AM							
	AN							
	AO							
	AP							

OTHER PRIOR ART (Including Author, Title, Date, Pertinent Pages, Etc.)

MJ	AR	GSM 06.10, "GSM Full Rate Speech Transcoding", Technical Rep. Vers. 3.2, ETSI/GSM, February 1992.
MJ	AS	P. Vary et al., "Speech Codec for European Mobile Radio System", Proc, ICASSP-88, p. 227, April 1988.
	AT	

EXAMINER	DATE CONSIDERED
M. Jung	9/16/97

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609; Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

A27960119947L

ETSI/TC SMG
Released by : ETSI/PT 12
Release date: February 1992

RELEASE NOTE

Recommendation GSM 06.10

GSM Full Rate Speech Transcoding

Previously distributed version : 3.2.0 (Release 1/90)
New Released version February 92 : 3.2.0 (Release 92, Phase 1)

1. Reason for changes

No changes since the previously distributed version.

GSM recommendation: 06.10

Title: GSM full rate speech transcoding

Date: February 1992

List of contents:

NOTE: This Recommendation is a reproduction of recommendation T/L/03/11 "13 kbit/s Regular Pulse Excitation - Long Term Prediction - Linear Predictive Coder for use in the Pan-European Digital Mobile Radio System".

Floppy disks containing the digital test sequences described in section 5 can be distributed by ETSI Secretariat on request.

The contact information of the ETSI secretariat is:

ETSI

B.P. 152

F 06561 Valbonne Cedex

France

Tel +33 92 94 42 00

Fax +33 93 65 47 16

<u>Language of original</u>: English

<u>Number of pages</u>: 93

Detailed list of contents
****

1. GENERAL

1.1. SCOPE

The transcoding procedure specified in this recommendation is applicable for the full-rate traffic channel (TCH) in the Pan-European Digital Mobile Radio (DMR) system. The use of this transcoding scheme for other applications has not been considered.

In recommendation GSM 06.01, a reference configuration for the speech transmission chain of the Pan-European DMR system is shown. According to this reference configuration, the speech encoder takes its input as a 13 bit uniform PCM signal either from the audio part of the mobile station or on the network side, from the PSTN via an 8 bit/A-law to 13 bit uniform PCM conversion. The encoded speech at the output of the speech encoder is delivered to a channel encoder unit which is specified in Rec.GSM 05.03. In the receive direction, the inverse operations take place.

This recommendation describes the detailed mapping between input blocks of 160 speech samples in 13 bit uniform PCM format to encoded blocks of 260 bits and from encoded blocks of 260 bits to output blocks of 160 reconstructed speech samples. The sampling rate is 8000 sample/s leading to an average bit rate for the encoded bit stream of 13 kbit/s. The coding scheme is the so-called Regular Pulse Excitation - Long Term prediction - Linear Predictive Coder, here-after referred to as RPE-LTP.

The recommendation also specifies the conversion between A-law PCM and 13 bit uniform PCM. Performance requirements for the audio input and output parts are included only to the extent that they affect the transcoder performance. The recommendation also describes the codec down to the bit level, thus enabling the verification of compliance to the recommendation to a high degree of confidence by use of a set of digital test sequences. These test sequences are also described and are available on floppy disks.

1.2. OUTLINE DESCRIPTION

The recommendation is structured as follows:

Section 1.3 contains a functional description of the audio parts including the A/D and D/A functions. Section 1.4 describes the conversion between 13 bit uniform and 8 bit A-law samples. Sections 1.5 and 1.6 present a simplified description of the principles of the RPE-LTP encoding and decoding process respectively. In section 1.7, the sequence and subjective importance of encoded parameters are given.

Section 2 deals with the transmission characteristics of the audio parts that are relevant for the performance of the RPE-LTP codec. Some transmission characteristics of the RPE-LTP codec are also specified in section 2. Section 3 presents the functional description of the RPE-LTP coding and decoding procedures, whereas section 4 describes the computational details of the algorithm. Procedures for the verification of the correct functioning of the RPE-LTP are described in section 5.

Performance and network aspects of the RPE-LTP codec are contained in annex 1.

1.3. FUNCTIONAL DESCRIPTION OF AUDIO PARTS

The analogue-to-digital and digital-to-analogue conversion will in principle comprise the following elements:

1) Analogue to uniform digital

- microphone,
- input level adjustment device,
- input anti-aliasing filter,
- sample-hold device sampling at 8 kHz,
- analogue-to-uniform digital conversion to 13 bits representation.

The uniform format shall be represented in two's complement.

2) Uniform digital to analogue

- conversion from 13 bit /8kHz uniform PCM to analogue,
- a hold device,
- reconstruction filter including x/sin x correction,
- output level adjustment device,
- earphone or loudspeaker.

In the terminal equipment, the A/D function may be achieved either

- by direct conversion to 13 bit uniform PCM format.
- or by conversion to 8 bit/A-law companded format, based on a standard A-law codec/filter according to CCITT rec. G.711/714, followed by the 8-bit to 13-bit conversion according to the procedure specified in section 1.4.

For the D/A operation, the inverse operations take place.

In the latter case it should be noted that the specifications in CCITT recommendation G.714 are concerned with PCM equipment located in the central parts of the network. When used in the terminal equipment, this specification does not on its own ensure sufficient out-of-band attenuation.

The specification of out-of-band signals is defined in section 2 between the acoustic signal and the digital interface to take into account that the filtering in the terminal can be achieved both by electronic and acoustical design.

1.4. PCM FORMAT CONVERSION

The conversion between 8 bit A-law companded format and the 13-bit uniform format shall be as defined in CCITT Recommendation G.721, section 4.2.1, sub-block EXPAND and section 4.2.7, sub-block COMPRESS. The parameter LAW = 1 should be used.

1.5. PRINCIPLES OF THE RPE-LTP ENCODER

A simplified block diagram of the RPE-LTP encoder is shown in Fig 1.1. In this diagram the coding and quantization functions are not shown explicitly.

The input speech frame, consisting of 160 signal samples (uniform 13 bit PCM samples), is first pre-processed to produce an offset-free signal, which is then subjected to a first order pre-emphasis filter. The 160 samples obtained are then analyzed to determine the coefficients for the short term analysis filter (LPC analysis). These parameters are then used for the filtering of the same 160 samples. The result is 160 samples of the short term residual signal. The filter parameters, termed reflection coefficients, are transformed to log.area ratios, LARs, before transmission.

For the following operations, the speech frame is divided into 4 sub-frames with 40 samples of the short term residual signal in each. Each sub-frame is processed blockwise by the subsequent functional elements.

Before the processing of each sub-block of 40 short term residual samples, the parame-ters of the long term analysis filter, the LTP lag and the LTP gain, are estimated and updated in the LTP analysis block, on the basis of the current sub-block of the present and a stored sequence of the 120 previous reconstructed short term residual samples.

A block of 40 long term residual signal samples is obtained by subtracting 40 estimates of the short term residual signal from the short term residual signal itself. The resulting block of 40 long term residual samples is fed to the Regular Pulse Excitation analysis which performs the basic compression function of the algorithm.

As a result of the RPE-analysis, the block of 40 input long term residual samples are represented by one of 4 candidate sub-sequences of 13 pulses each. The subsequence selected is identified by the RPE grid position (M). The 13 RPE pulses are encoded using Adaptive Pulse Code Modulation (APCM) with estimation of the sub-block amplitude which is transmitted to the decoder as side information.

The RPE parameters are also fed to a local RPE decoding and reconstruction module which produces a block of 40 samples of the quantized version of the long term residual signal.

By adding these 40 quantized samples of the long term residual to the previous block of short term residual signal estimates, a reconstructed version of the current short term residual signal is obtained.

The block of reconstructed short term residual signal samples is then fed to the long term analysis filter which produces the new block of 40 short term residual signal estimates to be used for the next sub-block thereby completing the feedback loop.

1.6. PRINCIPLES OF THE RPE-LTP DECODER

The simplified block diagram of the RPE-LTP decoder is shown in fig 1.2. The decoder includes the same structure as the feed-back loop of the encoder. In error-free transmission, the output of this stage will be the reconstructed short term residual samples. These samples are then applied to the short term synthesis filter followed by the de-emphasis filter resulting in the reconstructed speech signal samples.

1.7. SEQUENCE AND SUBJECTIVE IMPORTANCE OF ENCODED PARAMETERS

As indicated in fig 1.1 the three different groups of data are produced by the encoder are:

- the short term filter parameters,
- the Long Term Prediction (LTP) parameters
- the RPE parameters

The encoder will produce this information in a unique sequence and format, and the decoder must receive the same information in the same way. In table 1.1, the sequence of output bits b1 to b260 and the bit allocation for each parameter is shown.

The different parameters of the encoded speech and their individual bits have unequal importance with respect to subjective quality. Before being submitted to the channel encoder, the bits have to be rearranged in the sequence importance as given in table 1.2. The ranking has been determined by subjective testing and the procedure used is described in annex 2.

Parameter	Parameter number	Parameter name	Var. name	Number of bits	Bit no. (LSB-MSB)
	1		LAR 1	6	b1 - b6
	2		LAR 2	6	b7 - b12
FILTER	3	Log. Area	LAR 3	5	b13 - b17
	4	ratios	LAR 4	5	b18 - b22
PARAMETERS	5	1 - 8	LAR 5	4	b23 - b26
	6		LAR 6	4	b27 - b30
	7		LAR 7	3	b31 - b33
	8		LAR 8	3	b34 - b36
Sub-frame no.1					
LTP	9	LTP lag	N1	7	b37 - b43
PARAMETERS	10	LTP gain	b1	2	b44 - b45
	11	RPE grid position	M1	2	b46 - b47
RPE	12	Block amplitude	Xmax1	6	b48 - b53
PARAMETERS	13	RPE-pulse no.1	x1(0)	3	b54 - b56
	14	RPE-pulse no.2	x1(1)	3	b57 - b59
	..	...			...
	25	RPE-pulse no.13	x1(12)	3	b90 - b92
Sub-frame no.2					
LTP	26	LTP lag	N2	7	b93 - b99
PARAMETERS	27	LTP gain	b2	2	b100- b101
	28	RPE grid position	M2	2	b102- b103
RPE	29	Block amplitude	Xmax2	6	b104- b109
PARAMETERS	30	RPE-pulse no.1	x2(0)	3	b110- b112
	31	RPE-pulse no.2	x2(1)	3	b113- b115
	..	...			...
	42	RPE-pulse no.13	x2(12)	3	b146- b148

Table 1.1a. Encoder output parameters in order of occurrence and bit allocation within the speech frame of 260 bits/20 ms
--

Sub-frame no.3

LTP	43	LTP lag	N3	7	b149- b155
PARAMETERS	44	LTP gain	b3	2	b156- b157
	45	RPE grid position	M3	2	b158- b159
RPE	46	Block amplitude	Xmax3	6	b160- b165
PARAMETERS	47	RPE-pulse no.1	x3(0)	3	b166- b168
	48	RPE-pulse no.2	x3(1)	3	b169- b171
	..	...			...
	59	RPE-pulse no.13	x3(12)	3	b202- b204

Sub-frame no.4

LTP	60	LTP lag	N4	7	b205- b211
PARAMETERS	61	LTP gain	b4	2	b212- b213
	62	RPE grid position	M4	2	b214- b215
RPE	63	Block amplitude	Xmax4	6	b216- b221
PARAMETERS	64	RPE-pulse no.1	x4(0)	3	b222- b224
	65	RPE-pulse no.2	x4(1)	3	b225- b227
	..	...			...
	76	RPE-pulse no.13	x4(12)	3	b258- b260

Table 1.1b. Encoder output parameters in order of occurrence and bit allocation within the speech frame of 260 bits/20 ms

--

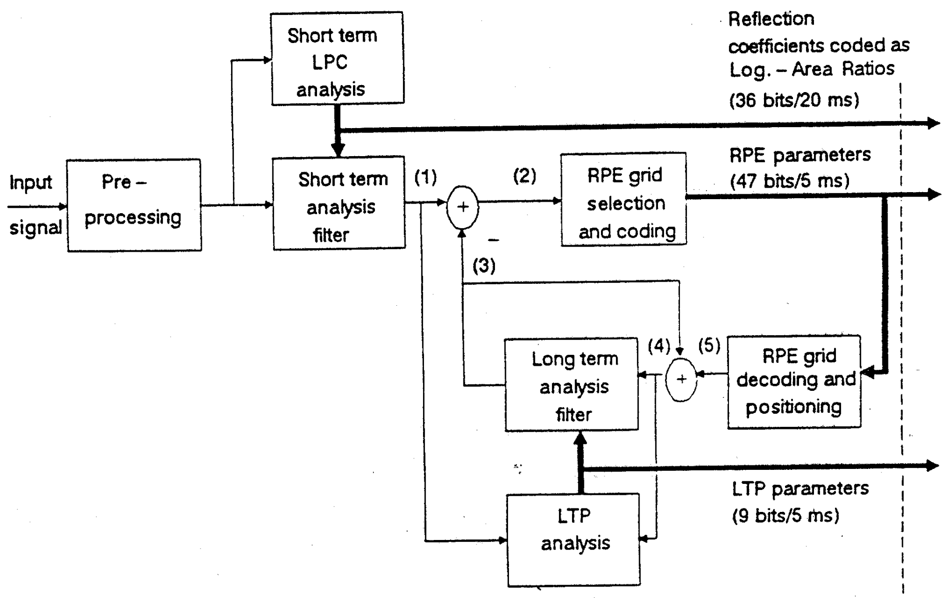
Order of bit importance	Parameter name	Parameter number	Bit number
Class Ia			
1	Log.area ratio 1	1	b6
2,3,4,5	Block amplitude	12,29,46,63	b53,b109,b165,b221
6	Log.area ratio 1	1	b5
7	Log.area ratio 2	2	b12
8	Log.area ratio 3	3	b17
.	Log.area ratio 1	1	b4
.	Log.area ratio 2	2	b11
	Log.area ratio 3	3	b16
	Log.area ratio 4	4	b22
	LTP lag	9,26,43,60	b43,b99,b155,b211
	Block amplitude	12,29,46,63	b52,b108,b164,b220
	Log.area ratio 2,5,6	2,5,6	b10,b26,b30
	LTP lag	9,26,43,60	b42,b98,b154,b210
	LTP lag	9,26,43,60	b41,b97,b153,b209
	LTP lag	9,26,43,60	b40,b96,b152,b208
	LTP lag	9,26,43,60	b39,b95,b151,b207
	Block amplitude	12,29,46,63	b51,b107,b163,b219
	Log.area ratio 1	1	b3
.	Log.area ratio 4	4	b21
.	Log.area ratio 7	7	b33
..49,50	LTP lag	9,26,43,60	b38,b94,b150,b206
Class Ib			
51,52	Log.area ratio 5,6	5,6	b25,b29
.	LTP gain	10,27,44,61	b45,b101,b157,b213
.	LTP lag	9,26,43,60	b37,b93,b149,b205
	Grid position	11,28,45,62	b47,b103,b159,b215
	Log.area ratio 1	1	b2
	Log.area ratio 2,3,8,4	2,3,8,4	b9,b15,b36,b20
	Log.area ratio 5,7	5,7	b24,b32
	LTP gain	10,27,44,61	b44,b100,b156,b212
	Block amplitude	12,29,46,63	b50,b106,b162,b218
	RPE pulses	13..25	b56,b59,..,b92
	RPE pulses	30..42	b112,b115,..,b148
	RPE pulses	47..59	b168,b171,..,b204
	RPE pulses	64..76	b224,b227,..,b260
	Grid position	11,28,45,62	b46,b102,b158,b214
	Block amplitude	12,29,46,63	b49,b105,b161,b217
	RPE pulses	13..25	b55,b58,..,b91
.	RPE pulses	30..42	b111,b114,..,b147
.	RPE pulses	47..59	b167,b170,..,b203
...182	RPE pulses	64..67	b223,b226,b229,b232

Table 1.2a. Subjective importance of encoded bits (the parameter and bit numbers refer to table 1.1)

Class II			
183...	RPE pulses	68..76	b235,b238,..,b259
	Log.area ratio 1	1	b1
	Log.area ratio 2,3,6	2,3,6	b8,b14,b28
	Log.area ratio 7	7	b31
	Log.area ratio 8	8	b35
	Log.area ratio 8,3	8,3	b34,b13
	Log.area ratio 4	4	b19
	Log.area ratio 4,5	4,5	b18,b23
	Block amplitude	12,29,46,63	b48,b104,b160,b216
	RPE pulses	13..25	b54,b57,..,b90
	RPE pulses	30..42	b110,b113,..,b146
	RPE pulses	47..59	b166,b169,..,b202
	RPE pulses	64..76	b222,b225,..,b258
259,260	Log.area ratio 2,6	2,6	b7,b27

Table 1.2b. Subjective importance of encoded bits (the parameter and bit numbers refer to table 1.1)

NOTE: The subdivisions in table 1.2 indicate the border between protection classes Ia, Ib and II as defined in recommendation GSM 05.03.



(1) Short term residual
(2) Long term residual (40 samples)
(3) Short term residual estimate (40 samples)
(4) Reconstructed short term residual (40 samples)
(5) Quantized long term residual (40 samples)

Fig 1.1. Simplified block diagram of the RPE – LTP encoder

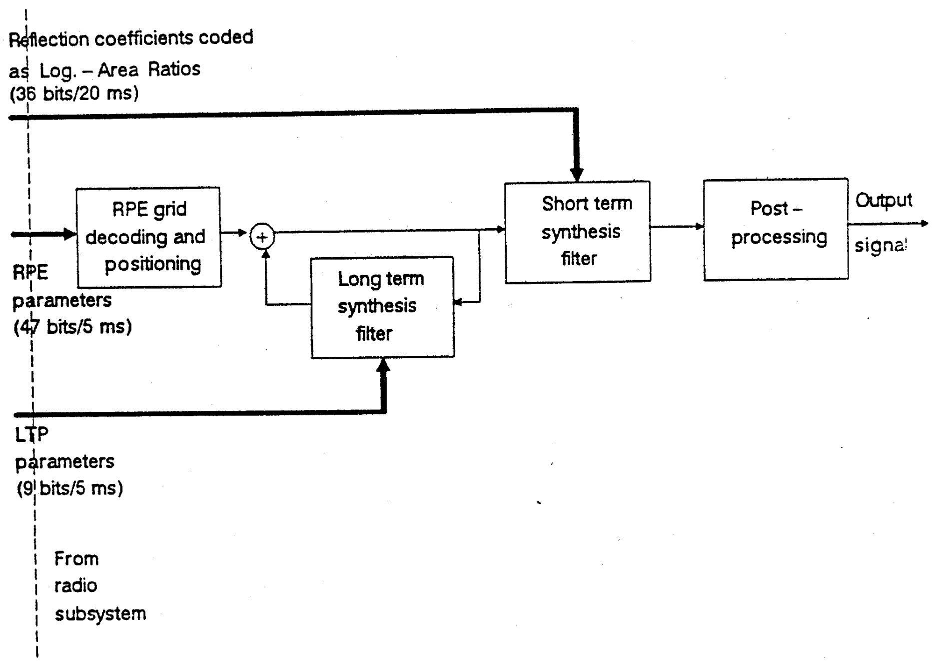


Fig 1.2. Simplified block diagram of the RPE – LTP decoder

2. TRANSMISSION CHARACTERISTICS

This section specifies the necessary performance characteristics of the audio parts for proper functioning of the speech trancoder. Some transmission performance characteristics of the RPE-LTP transcoder are also given to assist the designer of the speech transcoder function. The information given here is redundant and the detailed specifications are contained in recommendation GSM 11.10.

The performance characteristics are referred to the 13 bit uniform PCM interface.

NOTE: To simplify the verification of the specifications, the performance limits may be referred to an A-law measurement interface according to CCITT Rec-ommendation G.711. In this way, standard measuring equipments for PCM systems can be utilized for measurements. The relationship between the 13 bit format and the A-law companded shall follow the procedures defined in section 1.4.

2.1. PERFORMANCE CHARACTERISTICS OF THE ANALOGUE/DIGITAL INTERFACES

Concerning 1) discrimination against out-of-band signals (sending) and 2) spurious out-of-band signals (receiving), the same requirements as defined in ETSI standard TE 04-15 (digital telephone, candidate NET33) apply.

2.2. TRANSCODER DELAY

Consider a back to back configuration where the parameters generated by the encoder are delivered to the speech decoder as soon as they are available.

The transcoder delay is defined as the time interval between the instant a speech frame of 160 samples has been received at the encoder input and the instant the corresponding 160 reconstructed speech samples have been out-put by the speech decoder at an 8 kHz sample rate.

The theoretical minimum delay which can be achieved is 20 ms. The requirement is that the transcoder delay should be less than 30 ms.

3. FUNCTIONAL DESCRIPTION OF THE RPE-LTP CODEC

The block diagram of the RPE-LTP-coder is shown in fig 3.1. The individual blocks are described in the following sections.

3.1. FUNCTIONAL DESCRIPTION OF THE RPE-LTP ENCODER

The Preprocessing section of the RPE-LTP encoder comprises the following two sub-blocks:

* Offset compensation (3.1.1)
* Preemphasis (3.1.2)

The LPC analysis section of the RPE-LTP encoder comprises the following five sub-blocks:

* Segmentation (3.1.3)
* Auto-Correlation (3.1.4)
* Schur Recursion (3.1.5)
* Transformation of reflection coefficients to Log.-Area Ratios (3.1.6)
* Quantization and coding of Log.-Area Ratios (3.1.7)

The Short term analysis filtering section of the RPE-LTP comprises the following four sub-blocks:

* Decoding of the quantized Log.-Area Ratios (LARs) (3.1.8)
* Interpolation of Log.-Area Ratios (3.1.9)
* Transformation of Log.-Area Ratios into reflection coefficients (3.1.10)
* Short term analysis filtering (3.1.11)

The Long Term Predictor (LTP) section comprises 4 sub-blocks working on subsegments (3.1.12) of the short term residual samples.

* Calculation of LTP parameters (3.1.13)
* Coding of the LTP lags (3.1.14) and the LTP gains (3.1.15)
* Decoding of the LTP lags (3.1.14) and the LTP gains (3.1.15)
* Long term analysis filtering (3.1.16), and
  Long term synthesis filtering (3.1.17)

The RPE encoding section comprises five different sub-blocks:

* Weighting filter (3.1.18)
* Adaptive sample rate decimation by RPE grid selection (3.1.19)
* APCM quantization of the selected RPE sequence (3.1.20)
* APCM inverse quantization (3.1.21)
* RPE grid positioning (3.1.22)

PREPROCESSING SECTION

3.1.1. Offset compensation

Prior to the speech encoder an offset compensation,by a notch filter is applied in order to remove the offset of the input signal s_o to produce the offset-free signal s_{of}.

$$s_{of}(k) = s_o(k) - s_o(k-1) + alpha*s_{of}(k-1) \qquad (3.1.1)$$

$$alpha = 32735*2^{-15}$$

3.1.2. Preemphasis

The signal s_{of} is applied to a first order FIR preemphasis filter leading to the input signal s of the analysis section.

$$s(k) = s_{of}(k) - beta*s_{of}(k-1) \qquad (3.1.2)$$

$$beta= 28180*2^{-15}$$

LPC ANALYSIS SECTION

3.1.3. Segmentation

The speech signal s(k) is divided into non-overlapping frames having a length of T_0 = 20 ms (160 samples). A new LPC-analysis of order p=8 is performed for each frame.

3.1.4. Autocorrelation

The first p+1 = 9 values of the Auto-Correlation function are calculated by

$$ACF(k) = \sum_{i=k}^{159} s(i)s(i-k) \quad ,k = 0,1...,8 \qquad (3.2)$$

3.1.5. Schur Recursion

The reflection coefficients are calculated as shown in Fig 3.2 using the Schur Recursion algorithm. The term "reflection coefficient" comes from the theory of linear prediction of speech (LPC), where a vocal tract representation consisting of series of uniform cylindrical sections is assumed. Such a representation can be described by the reflection coefficents or the area ratios of connected sections.

3.1.6. Transformation of reflection coefficients to Log.-Area Ratios

The reflection coefficients r(i), (i=1..8), calculated by the Schur algorithm, are in the range

$$-1 <= r(i) <= + 1$$

Due to the favourable quantization characteristics, the reflection coefficients are converted into Log.-Area Ratios which are strictly defined as follows:

$$Logarea(i) = \log_{10} \left(\frac{1 + r(i)}{1 - r(i)} \right) \qquad (3.3)$$

Since it is the companding characteristic of this transformation that is of importance, the following segmented approximation is used.

```
          r(i)                       ;          |r(i)|  < 0.675
LAR(i) =  sign[r(i)]*[2|r(i)|-0.675] ; 0.675 <= |r(i)|  < 0.950
          sign[r(i)]*[8|r(i)|-6.375] ; 0.950 <= |r(i)| <= 1.000

                                                     (3.4)
```

with the result that instead of having to divide and obtain the logarithm of particular values, it is merely necessary to multiply, add and compare these values.

The following equation (3.5) gives the inverse transformation.

```
      LAR'(i)                            ; |LAR'(i)|<0.675
r'(i)=sign[LAR'(i)]*[0.500*|LAR'(i)|
      +0.337500]                         ; 0.675<=|LAR'(i)|<1.225
      sign[LAR'(i)]*[0.125*|LAR'(i)|
      +0.796875]                         ; 1.225<=|LAR'(i)|<=1.625
                                                     (3.5)
```

3.1.7. Quantization and coding of Log.-Area Ratios

The Log.-Area Ratios LAR(i) have different dynamic ranges and different asymmetric distribution densities. For this reason, the transformed coefficients LAR(i) are limited and quantized differently according to the following equation (3.6), with $LAR_c(i)$ denoting the quantized and integer coded version of LAR(i).

$$LAR_c(i) = Nint\{A(i)*LAR(i) + B(i)\} \qquad (3.6)$$

with

$$Nint\{z\} := int\{z+sign\{z\}*0.5\} \qquad (3.6a)$$

Function Nint defines the rounding to the nearest integer value, with the coefficients A(i), B(i), and different extreme values of $LAR_c(i)$ for each coefficient LAR(i) given in table 3.1.

LAR No i	A(i)	B(i)	Minimum $LAR_C(i)$	Maximum $LAR_C(i)$
1	20.000	0.000	-32	+31
2	20.000	0.000	-32	+31
3	20.000	4.000	-16	+15
4	20.000	-5.000	-16	+15
5	13.637	0.184	- 8	+ 7
6	15.000	-3.500	- 8	+ 7
7	8.334	-0.666	- 4	+ 3
8	8.824	-2.235	- 4	+ 3

Table 3.1. Quantization of the Log.-Area Ratios LAR(i)

SHORT-TERM ANALYSIS FILTERING SECTION

The current frame of the speech signal s is retained in memory until calculation of the LPC parameters LAR(i) is completed. The frame is then read out and fed to the short term analysis filter of order p=8. However, prior to the analysis filtering operation, the filter coefficients are decoded and preprocessed by interpolation.

3.1.8. Decoding of the quantized Log.-Area Ratios

In this block the quantized and coded Log.-Area Ratios ($LAR_C(i)$) are decoded according to equation (3.7).

$$LAR''(i) = (LAR_C(i) - B(i)) / A(i) \qquad (3.7)$$

3.1.9. Interpolation of Log.-Area Ratios

To avoid spurious transients which may occur if the filter coefficients are changed abruptly, two subsequent sets of Log.-Area Ratios are interpolated linearly. Within each frame of 160 analysed speech samples the short term analysis filter and the short term synthesis filter operate with four different sets of coefficients derived according to table 3.2.

k	$LAR'_J(i)$ =
0...12	$0.75*LAR''_{J-1}(i) + 0.25*LAR''_J(i)$
13...26	$0.50*LAR''_{J-1}(i) + 0.50*LAR''_J(i)$
27...39	$0.25*LAR''_{J-1}(i) + 0.75*LAR''_J(i)$
40..159	$LAR''_J(i)$

Table 3.2. Interpolation of LAR parameters (J=actual segment)

3.1.10. Transformation of Log.-Area Ratios into reflection coefficients

The reflection coefficients are finally determined using the inverse transformation according to equation (3.5).

3.1.11. Short Term Analysis Filtering

The Short term analysis filter is implemented according to the lattice structure depicted in fig 3.3.

$$d_0(k) = s(k) \qquad (3.8a)$$
$$u_0(k) = s(k) \qquad (3.8b)$$
$$d_i(k) = d_{i-1}(k) + r'_i*u_{i-1}(k-1) \quad \text{with } i=1,...8 \qquad (3.8c)$$
$$u_i(k) = u_{i-1}(k-1) + r'_i*d_{i-1}(k) \quad \text{with } i=1,...8 \qquad (3.8d)$$
$$d(k\ ) = d_8(k) \qquad (3.8e)$$

LONG-TERM PREDICTOR (LTP) SECTION

3.1.12. Sub-segmentation

Each input frame of the short term residual signal contains 160 samples, corresponding to 20 ms. The long term correlation is evaluated four times per frame, for each 5 ms subsegment. For convenience in the following, we note j=0,...,3 the sub-segment number, so that the samples pertaining to the j-th sub-segment of the residual signal are now denoted by $d(k_j+k)$ with j = 0,...,3; $k_j = k_0 + j*40$ and k = 0,...,39 where k_0 corresponds to the first value of the current frame.

3.1.13. Calculation of the LTP parameters

For each of the four sub-segments a long term correlation lag N_j, ($j=0,\ldots,3$), and an associated gain factor b_j, ($j=0,\ldots,3$) are determined. For each sub-segment, the determination of these parameters is implemented in three steps.

1) The first step is the evaluation of the cross-correlation R_j(lambda) of the current sub-segment of short term residual signal $d(k_j+i)$,($i=0,\ldots,39$) and the previous samples of the reconstructed short term residual signal $d'(k_j+i)$, ($i=-120,\ldots,-1$):

$$R_j(\text{lambda}) = \sum_{i=0}^{39} d(k_j+i) * d'(k_j+i-\text{lambda}); \quad \begin{array}{l} j = 0,\ldots 3 \\ k_j = k_0 + j*40 \\ \text{lambda} = 40,\ldots,120 \end{array} \qquad (3.9)$$

The cross-correlation is evaluated for lags lambda greater than or equal to 40 and less than or equal to 120, ie corresponding to samples outside the current sub-segment and not delayed by more than two sub-segments.

2) The second step is to find the position N_j of the peak of the cross-correlation function within this interval:

$$R_j(N_j) = \max \{ R_j(\text{lambda});\ \text{lambda} = 40..120 \}; \quad j = 0,\ldots,3 \qquad (3.10)$$

3) The third step is the evaluation of the gain factor b_j according to:

$$b_j = R_j(N_j) \,/\, S_j(N_j); \qquad j = 0,\ldots,3 \qquad (3.11)$$

with

$$S_j(N_j) = \sum_{i=0}^{39} d'^2 (k_j+i-N_j); \quad j = 0,\ldots,3 \qquad (3.12)$$

It is clear that the last 120 samples of the reconstructed short term residual signal $d'(k_j+i),(i=-120,\ldots,-1)$ must be retained until the next sub-segment so as to allow the evaluation of the relations (3.9),...,(3.12).

3.1.14. Coding/Decoding of the LTP lags

The long term correlation lags $N_j,(j=0,\ldots,3)$ can have values in the range (40,...,120), and so must be coded using 7 bits with:

$$N_{cj} = N_j; \qquad j = 0,\ldots,3 \qquad (3.13)$$

At the receiving end, assuming an error free transmission, the decoding of these values will restore the actual lags:

$$N_j' = N_{cj}; \qquad j = 0,\ldots,3 \qquad (3.14)$$

3.1.15. Coding/Decoding of the LTP gains

The long term prediction gains $b_j,(j=0,\ldots,3)$ are encoded with 2 bits each, according to the following algorithm:

$$\begin{array}{lll} \text{if } b_j \le DLB(i) & \text{then } b_{cj} = 0; & i=0 \\ \text{if } DLB(i-1) < b_j \le DLB(i) & \text{then } b_{cj} = i; & i=1,2 \qquad (3.15) \\ \text{if } DLB(i-1) < b_j & \text{then } b_{cj} = 3; & i=3 \end{array}$$

where $DLB(i),(i=0,\ldots,2)$ denotes the decision levels of the quantizer, and b_{cj} represents the coded gain value. Decision levels and quantizing levels are given in table 3.3.

i	Decision level DLB(i)	Quantizing level QLB(i)
0	0.2	0.10
1	0.5	0.35
2	0.8	0.65
3		1.00

Table 3.3. Quantization table for the LTP gain

The decoding rule is implemented according to:

$$b_j' = QLB(b_{cj}) \; ; \; j = 0,...,3 \qquad (3.16)$$

where QLB(i),(i=0,...,3) denotes the quantizing levels, and b_j' represents the decoded gain value (see table 3.3).

3.1.16. Long term analysis filtering

The short term residual signal $d(k_0+k)$,(k=0,...,159) is processed by sub-segments of 40 samples. From each of the four sub-segments (j=0,...,3) of short term residual samples, denoted here $d(k_j+k)$, (k=0,...,39), an estimate $d"(k_j+k)$, (k=0,...,39) of the signal is subtracted to give the long term residual signal $e(k_j+k)$, (k=0,...,39) (see fig 3.1):

$$e(k_j+k) = d(k_j+k) - d"(k_j+k) \quad ; \quad \begin{array}{l} j = 0,...,3 \\ k = 0,...,39 \\ k_j = k_0 + j*40 \end{array} \qquad (3.17)$$

Prior to this subtraction, the estimated samples $d"(k_j+k)$ are computed from the previously reconstructed short term residual samples d', adjusted to the current sub-segment LTP lag N_j' and weighted with the sub-segment LTP gain b_j':

$$d"(k_j+k) = b_j'*d'(k_j+k-N_j') \quad ; \quad \begin{array}{l} j = 0,...,3 \\ k = 0,...,39 \\ k_j = k_0 + j*40 \end{array} \qquad (3.18)$$

3.1.17. Long term synthesis filtering

The reconstructed long term residual signal $e'(k_0+k)$,(k=0,...,159) is processed by sub-segments of 40 samples. To each sub-segment, denoted here $e'(k_j+k)$, (k=0,...,39), the estimate $d"(k_j+k)$, (k=0,...,39) of the signal is added to give the reconstructed short term residual signal $d'(k_j+k)$,(k=0,...,39):

$$d'(k_j+k) = e'(k_j+k) + d"(k_j+k) \quad ; \quad \begin{array}{l} j = 0,...,3 \\ k = 0,...,39 \\ k_j = k_0 + j*40 \end{array} \qquad (3.19)$$

RPE ENCODING SECTION

3.1.18. Weighting Filter

A FIR 'block filter' algorithm is applied to each sub-segment by convolving 40 samples e(k) with the impulse response H(i) ; i=0,...,10 (see table 3.4).

i	5	4 (6)	3 (7)	2 (8)	1 (9)	0 (10)
$H(i)*2^{13}$	8192	5741	2054	0	-374	-134

|H(Omega=0)| = 2.779;

Table 3.4. Impulse response of block filter (weighting filter)

The conventional convolution of a sequence having 40 samples with an 11-tap impulse response would produce 40+11-1=50 samples. In contrast to this, the 'block filter' algorithm produces the 40 central samples of the conventional convolution operation. For notational convenience the block filtered version of each sub-segment is denoted by x(k), k=0,...,39.

$$x(k) = \sum_{i=0}^{10} H(i) * e(k+5-i) \qquad \text{with } k = 0,...,39 \qquad (3.20)$$

<u>NOTE</u>: e(k+5-i) = 0 for k+5-i<0 and k+5-i>39.

3.1.19. Adaptive sample rate decimation by RPE grid selection

For the next step, the filtered signal x is down-sampled by a ratio of 3 resulting in 3 interleaved sequences of lengths 14, 13 and 13, which are split up again into 4 sub-sequences x_m of length 13:

$$x_m(i) = x(k_j+m+3*i) \qquad ; \quad i = 0,...,12 \quad m = 0,...,3 \qquad (3.21)$$

with m denoting the position of the decimation grid. According to the explicit solution of the RPE mean squared error criterion, the optimum candidate sub-sequence x_M is selected which is the one with the maximum energy

$$E_M = \max_{m} \sum_{i=0}^{12} x_m^2(i) \qquad ; \quad m = 0,\ldots,3 \qquad (3.22)$$

The optimum grid position M is coded as M_c with 2 bits.

3.1.20. APCM quantization of the selected RPE sequence

The selected sub-sequence $x_M(i)$ (RPE sequence) is quantized, applying APCM (Adaptive Pulse Code Modulation). For each RPE sequence consisting of a set of 13 samples $x_M(i)$,the maximum x_{max} of the absolute values $|x_M(i)|$ is selected and quantized logarithmically with 6 bits as x_{maxc} as given in table 3.5.

x_{max}	x'_{max}	x_{maxc}	x_{max}	x'_{max}	x_{maxc}
0 .. 31	31	0	2048 .. 2303	2303	32
32 .. 63	63	1	2304 .. 2559	2559	33
64 .. 95	95	2	2560 .. 2815	2815	34
96 .. 127	127	3	2816 .. 3071	3071	35
128 .. 159	159	4	3072 .. 3327	3327	36
160 .. 191	191	5	3328 .. 3583	3583	37
192 .. 223	223	6	3584 .. 3839	3839	38
224 .. 255	255	7	3840 .. 4095	4095	39
256 .. 287	287	8	4096 .. 4607	4607	40
288 .. 319	319	9	4608 .. 5119	5119	41
320 .. 351	351	10	5120 .. 5631	5631	42
352 .. 383	383	11	5632 .. 6143	6143	43
384 .. 415	415	12	6144 .. 6655	6655	44
416 .. 447	447	13	6656 .. 7167	7167	45
448 .. 479	479	14	7168 .. 7679	7679	46
480 .. 511	511	15	7680 .. 8191	8191	47
512 .. 575	575	16	8192 .. 9215	9215	48
576 .. 639	639	17	9216 .. 10239	10239	49
640 .. 703	703	18	10240 .. 11263	11263	50
704 .. 767	767	19	11264 .. 12287	12287	51
768 .. 831	831	20	12288 .. 13311	13311	52
832 .. 895	895	21	13312 .. 14335	14335	53
896 .. 959	959	22	14336 .. 15359	15359	54
960 .. 1023	1023	23	15360 .. 16383	16383	55
1024 .. 1151	1151	24	16384 .. 18431	18431	56
1152 .. 1279	1279	25	18432 .. 20479	20479	57
1280 .. 1407	1407	26	20480 .. 22527	22527	58
1408 .. 1535	1535	27	22528 .. 24575	24575	59
1536 .. 1663	1663	28	24576 .. 26623	26623	60
1664 .. 1791	1791	29	26624 .. 28671	28671	61
1792 .. 1919	1919	30	28672 .. 30719	30719	62
1920 .. 2047	2047	31	30720 .. 32767	32767	63

Table 3.5. Quantization of the block maximum x_{max}

For the normalization, the 13 samples are divided by the decoded version x'_{max} of the block maximum. Finally, the normalized samples

$$x'(i) = x_M(i)/x'_{max} \; ; \qquad i=0,\ldots,12 \qquad (3.23)$$

are quantized uniformly with three bits to $x_{MC}(i)$ as given in table 3.6.

$x'*2^{15}$ (Interval-limits)	$\underline{x}_M'*2^{15}$	x_{Mc} (Channel)
-32768 ... -24577	-28672	0 = 000
-24576 ... -16385	-20480	1 = 001
-16384 ... -8193	-12288	2 = 010
-8192 ... -1	-4096	3 = 011
0 ... 8191	4096	4 = 100
8192 ... 16383	12288	5 = 101
16384 ... 24575	20480	6 = 110
24576 ... 32767	28672	7 = 111

Table 3.6. Quantization of the normalized RPE-samples

3.1.21. APCM inverse quantization

The $x_{Mc}(i)$ are decoded to $\underline{x}_M'(i)$ and denormalized using the decoded value x'_{maxc} leading to the decoded sub-sequence $x'_M(i)$.

3.1.22. RPE grid positioning

The quantized sub-sequence is upsampled by a ratio of 3 by inserting zero values according to the grid position given with Mc.

3.2. DECODER

The decoder comprises the following 4 sections. Most of the sub-blocks are also needed in the encoder and have been described already. Only the short term tynthesis filter and the deemphasis filter are added in the decoder as new sub-blocks.

* RPE decoding section (3.2.1)
* Long Term Prediction section (3.2.2)
* Short term synthesis filtering section (3.2.3)
* Postprocessing (3.2.4)

The complete block diagram for the decoder is shown in fig 3.4. The variables and parameters of the decoder are marked by the index r to distinguish the received values from the encoder values.

3.2.1. RPE decoding section

The input signal of the long term synthesis filter (reconstruction of the long term residual signal) is formed by decoding and denormalizing the RPE-samples (APCM inverse quantization - 3.1.21) and by placing them in the correct time position (RPE grid positioning - 3.1.22). At this stage, the sampling frequency is increased by a factor of 3 by inserting the appropriate number of intermediate zero-valued samples.

3.2.2. Long Term Prediction section

The the reconstructed long term residual signal e_r' is applied to the long term synthesis filter (see 3.1.16 and 3.1.17) which produces the reconstructed short term residual signal d_r' for the short term synthesiser.

3.2.3. Short term synthesis filtering section

The coefficients of the short term synthesis filter (see fig 3.5) are reconstructed applying the identical procedure to that in the encoder (3.1.8 - 3.1.10). The short term synthesis filter is implemented according to the lattice structure depicted in fig 3.5.

$$s_{r(0)}(k) = d_r'(k) \tag{3.24a}$$

$$s_{r(i)}(k) = s_{r(i-1)}(k) - r_r'{}_{(9-i)} * v_{8-i}(k-1); \quad i=1,\ldots,8 \tag{3.24b}$$

$$v_{9-i}(k) = v_{8-i}(k-1) + r_r'{}_{(9-i)} * s_{r(i)}(k); \quad i=1,\ldots,8 \tag{3.24c}$$

$$s_r'(k) = s_{r(8)}(k) \tag{3.24d}$$

$$v_0(k) = s_{r(8)}(k) \tag{3.24e}$$

3.2.4. Postprocessing

The output of the synthesis filter $s_r(k)$ is fed into the IIR-deemphasis filter leading to the output signal s_{ro}.

$$s_{ro}(k) = s_r(k) + beta*s_{ro}(k-1) \;; \quad beta= 28180*2^{-15} \tag{3.25}$$

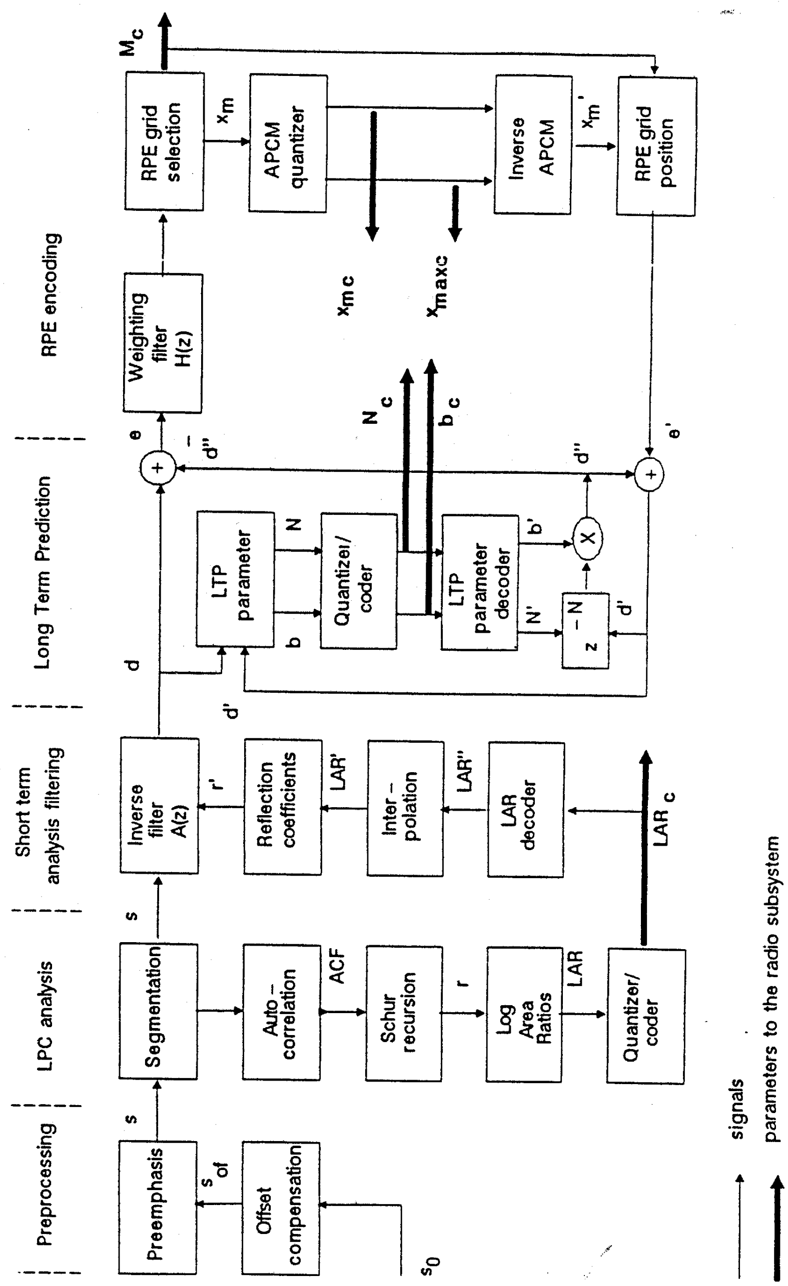


Fig 3.1. Block diagram of the RPE – LTP encoder

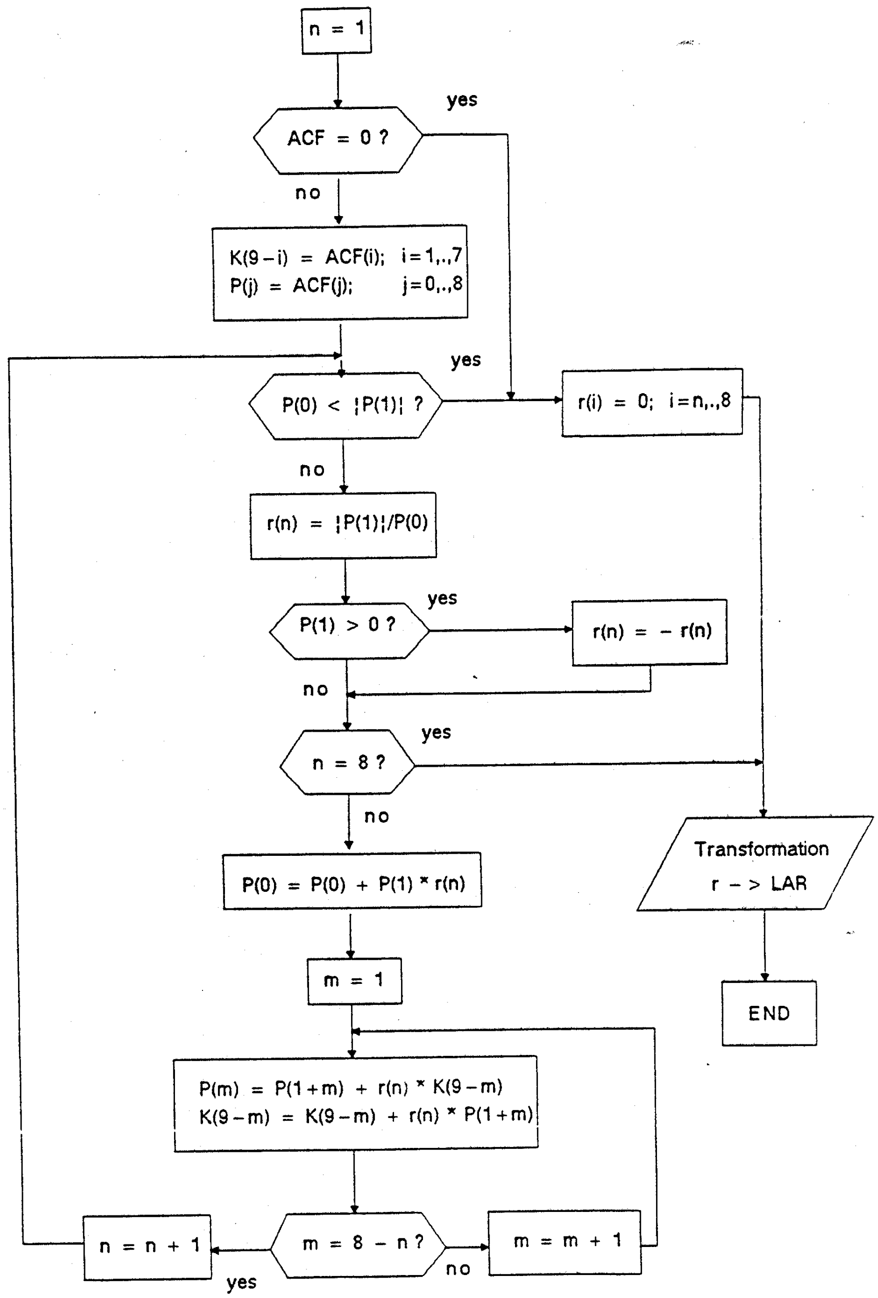


Fig 3.[illegible] LPC analysis using Schur recursion

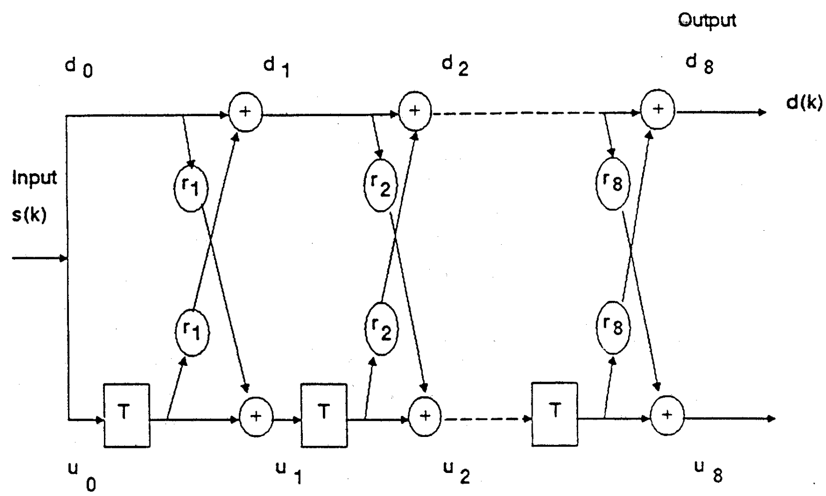


Fig 3.3. Short term analysis filter

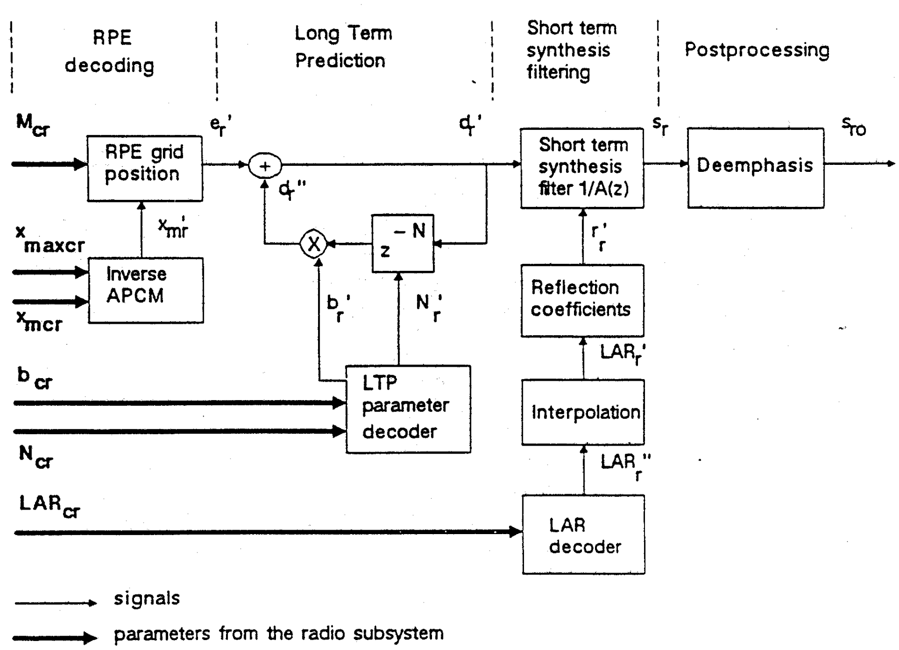


Fig 3.4. Block diagram of the RPE – LTP decoder

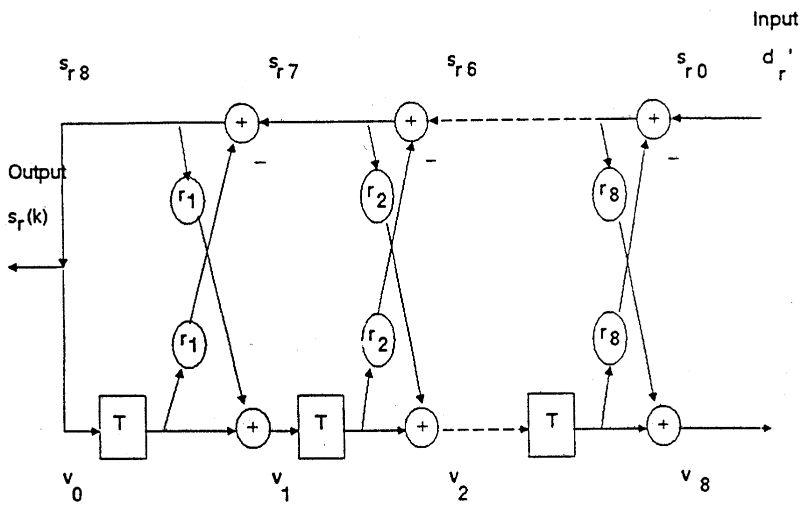


Fig 3.5. Short term synthesis filter

4. COMPUTATIONAL DETAILS OF THE RPE-LTP CODEC

4.1. DATA REPRESENTATION AND ARITHMETIC OPERATIONS

Only two types of variables are used along the implementation of the RPE-LTP algorithm in fixed point arithmetic. These two types are:

Integer on 16 bits;

Long integer on 32 bits;

This assumption simplifies the detailed description and allows the maximum reach of precision.

In different places of the recommendation, different scaling factors are used according to different operations. To help the reader in the comparison of corresponding floating point and fixed point values given in section 3 and 4 comments of the format:

```
/* var  = integer(  real_var * scalefactor ) */
```

are used at several points of section 4. var is the rounded fixed point representation of the floating point representation of var (real_var) using the given scaling factor.

In the description, input signal samples, coded parameters and output signal samples are represented by 16 bit words. At the receiving part it must therefore be ensured that only valid bits (13 bits for samples signal and two to seven bits for coded parameters) are used. In verification tests, the testing system may introduce random bit at non valid places inside these samples (3 LSBs) or parameters (MSBs) to test this function. In the digital test sequences all non valid bits are set to 0.

The following part of this section describes the required set of arithmetic operations to implement the RPE-LTP algorithm in fixed point.

For arithmetics operations or variables with a long integer type (32 bit) a prefix L_ is used in order to distinguish them from the 16 bit variables or arithmetic operations.

All the names of the variables are identical to those of the functional description of the RPE-LTP Codec (Section 3) but variables like x', x'' are respectively called:

```
x' -----> xp
x''-----> xpp
```

in order to avoid any confusing notation.

NOTE: The x',x" variables are examples but are not used within the following description.

The following notations are used in the arithmetic operations:

Square brackets ([..]) are used for arrays and when needed, the starting index and the ending index are put inside the bracket. For example x[0..159] means that x is an array of 160 words of 16 bits with beginning index 0 and ending index 159 and x[k] is an element of the array x[0..159].

All functions' names are underlined. For example add(x, y) means that we perform the addition of x and y.

<< n : denotes a n-bit arithmetic shift left operation (zero fill) on variables of type short or long; if n is less than 0, this operation becomes an arithmetic right shift of -n.

>> n : denotes a n-bit arithmetic right shift operation (sign extension) on variables of type short or long; if n is less than 0, this operation becomes an arithmetic left shift of -n (zero fill).

a > b : denotes the "greater than" condition;

a >= b : denotes the "greater than or equal" condition;

a < b : denotes the "less than" condition;

a <= b : denotes the "less than or equal" condition;

a == b : denotes the "equal to" condition.

The basic structure of the FOR-NEXT loop is used in this description for loop computation; the declaration is:

```
|== FOR k= start to end:
|    inner computation;
|== NEXT k:
```

Also the IF.. ELSE IF structure is used throughout this detailed description. The basic structure is:

```
IF (condition1) THEN statement1;
   ELSE IF ( condition2) THEN statement2;
        ELSE IF ( condition3) THEN statement3;
```

The word EXIT is used to exit immediately from a procedure.

The following arithmetic operations are defined:

```
add( var1, var2)     : performs the addition (var1+var2) with
                     : overflow control and saturation; the result
                     : is set at +32767 when overflow occurs or at
                     : -32768 when underflow occurs.

sub( var1, var2)     : performs the subtraction (var1-var2) with
                     : overflow control and saturation; the result
                     : is set at +32767 when overflow occurs or at
                     : -32768 when underflow occurs.

mult( var1, var2)  : performs the multiplication of var1 by var2
                   : and gives a 16 bits result which is scaled ie
                   : mult( var1,var2 ) = (var1 times var2) >> 15
                   : and mult( -32768, -32768 ) = 32767

mult_r( var1, var2): same as mult but with rounding ie
                   : mult_r( var1, var2 ) =
                   : ( (var1 times var2) + 16384 ) >> 15
                   : and mult_r( -32768, -32768 ) = 32767

abs( var1 )      : absolute value of var1; abs(-32768) = 32767

div( var1, var2)   : div produces a  result which is the
                   : fractional integer division of var1 by var2;
                   : var1 and var2 must be positive and var2 must
                   : be greater or equal to var1; The result is
                   : positive (leading bit equal to 0) and
                   : truncated to 16 bits. if var1 == var2 then
                   : div( var1, var2 ) = 32767

L_mult(var1, var2) : L_mult is a 32 bit result for the
                   : multiplication of var1 times var2 with a one
                   : bit shift left. L_mult( var1, var2 ) =
                   | ( var1 times var2 ) << 1. The condition
                   | L_mult( -32768, -32768 ) does not occur in
                   | the algorithm.
```

```
L_add(L_var1, L_var2): 32 bits addition of two 32 bits variables
                     : (L_var1 + L_var2) with overflow control and
                     : saturation; the result is set at 2147483647
                     : when overflow occurs and at -2147483648
                     : when underflow occurs.

L_sub(L_var1,L_var2): 32 bits subtraction of two 32 bits variables
                    : (L_var1 - L_var2) with overflow control and
                    : saturation; the result is set at 2147483647
                    : when overflow occurs and at -2147483648
                    : when underflow occurs.

norm( L_var1 )      : norm produces the number of left shifts
                    : needed to normalize the 32 bits variable
                    : L_var1 for positive values on the interval
                    : with  minimum of 1073741824 and maximum of
                    : 2147483647 and for negative values on the
                    : interval with  minimum of -2147483648 and
                    : maximum of -1073741824; in order to normalize
                    : the result, the following operation must be
                    : done: L_norm_var1 = L_var1 << norm(L_var1)

L_var2 = var1;       : deposit the 16 bits of var1 in the LSB 16
                     : bits of L_var2 with sign extension.

var2 = L_var1;       : extract the 16 LSB bits of L_var1 to put in
                     : var2.
```

When a constant is used in an operation on 32 bits, it must be first sign-extended on 32 bits.

4.2. FIXED POINT IMPLEMENTATION OF THE RPE-LTP CODER

The RPE-LTP coder works on a frame by frame basis. The length of the frame is equal to 160 samples. Some computations are done once per frame (analysis) and some others for each of the four sub-segments (40 samples).

In the following detailed description, procedure 4.2.0 to 4.2.10 are done once per frame to produce at the output of the coder the LARc[1..8] parameters which are the coded LAR coefficients and also to realize the inverse filtering operation for the entire frame (160 samples of signal d[0..159]). These parts produce at the output of the coder:

```
| LARc[1..8]     : Coded LAR coefficients

|--> These parameters are calculated and sent once per
     frame.
```

Procedure 4.2.11 to 4.2.18 are to be executed four times per frame. That means once for each sub-segment RPE-LTP analysis of 40 samples. These parts produce at the output of the coder:

```
| Nc          : LTP lag;

| bc          : Coded LTP gain;

| Mc          : RPE grid selection;

| xmaxc          : Coded maximum amplitude of the RPE
                   sequence;

| xMc[0..12]     : Codes of the normalized RPE samples;

|--> These parameters are calculated and sent four times
     per frame.
```

PREPROCESSING SECTION

4.2.0. Scaling of the input variable

After A-law to linear conversion (or directly from the A to D converter) the following scaling is assumed for input to the RPE-LTP algorithm:

S.v.v.v.v.v.v.v.v.v.v.v.v.x.x.x (2's complement format).

Where S is the sign bit, v a valid bit, and x a "don t care" bit.

The original signal is called sop[..];

4.2.1. Downscaling of the input signal

```
|== FOR k=0 to 159:
|    so[k]  = sop[k] >> 3;
|    so[k]  = so[k] << 2;
|== NEXT k:
```

4.2.2. Offset compensation

This part implements a high-pass filter and requires extended arithmetic precision for the recursive part of this filter.

The input of this procedure is the array so[0..159] and the output the array sof[0..159].

```
|== FOR k = 0 to 159:
| Compute the non-recursive part.
|    s1 = sub( so[k], z1 );
|    z1 = so[k];
| Compute the recursive part.
|    L_s2 = s1;
|    L_s2 = L_s2 << 15;
```

```
| Execution of a 31 by 16 bits multiplication.
|     msp = L_z2 >> 15;
|     lsp = L_sub( L_z2, ( msp << 15 ) );
|     temp = mult_r( lsp, 32735 );
|     L_s2 = L_add( L_s2, temp );
|     L_z2 = L_add(  L_mult( msp, 32735 ) >> 1, L_s2 );
| Compute sof[k] with rounding.
|     sof[k] =  L_add( L_z2, 16384 )  >> 15;
|== NEXT k:
```

Keep z1 and L-z2 in memory for the next frame.

Initial value: z1=0; L_z2=0;

4.2.3. Preemphasis

```
|== FOR k=0 to 159:
|     s[k] = add( sof[k], mult_r( mp, -28180 ) );
|     mp = sof[k];
|== NEXT k:
```

Keep mp in memory for the next frame.

Initial value: mp=0;

LPC ANALYSIS SECTION

4.2.4. Autocorrelation

The goal is to compute the array L_ACF[k]. The signal s[i] must be scaled in order to avoid an overflow situation.

Dynamic scaling of the array s[0..159].

Search for the maximum.

```
smax = 0;
|== FOR k = 0 to 159:
|     temp = abs( s [k] );
|     IF ( temp > smax ) THEN smax = temp;
|== NEXT k;
```

Computation of the scaling factor.

```
IF ( smax == 0 ) THEN scalauto = 0;
  ELSE  scalauto = sub( 4, norm( smax << 16 )  );
```

Scaling of the array s[0..159].

```
IF ( scalauto > 0 ) THEN
                          | temp = 16384 >> sub( scalauto,1);
                          |== FOR k = 0 to 159:
                          |     s[k] = mult_r( s[k], temp);
                          |== NEXT k:
```

Compute the L-ACF[..].

```
|== FOR k=0 to 8:
|     L_ACF[k] = 0;
|==== FOR i=k to 159:
|       L_temp = L_mult( s[i], s[i-k] );
|       L_ACF[k] = L_add( L_ACF[k], L_temp );
|==== NEXT i:
|== NEXT k:
```

Rescaling of the array s[0..159].

```
        IF ( scalauto > 0 ) THEN
                                |== FOR k = 0 to 159:
                                |    s[k] = s[k] << scalauto;
                                |== NEXT k:
```

4.2.5. Computation of the reflection coefficients

Schur recursion with 16 bits arithmetic.

```
        IF( L_ACF[0] == 0 ) THEN
                                |== FOR i = 1 to 8:
                                |    r[i] = 0;
                                |== NEXT i:
                                |    EXIT; /continue with section
                                                 4.2.6/
        temp = norm( L_ACF[0] );
        |== FOR k=0 to 8:
        |    ACF[k] = ( L_ACF[k] << temp ) >> 16;
        |== NEXT k:
```

Initialize array P[..] and K[..] for the recursion.

```
        |== FOR i=1 to 7:
        |    K[9-i] = ACF[i];
        |== NEXT i:
```

```
|== FOR i=0 to 8:
|     P[i] = ACF[i];
|== NEXT i:
```

Compute reflection coefficients.

```
|== FOR n=1 to 8:
|     IF( P[0] < abs( P[1] ) ) THEN
|                                          |== FOR i = n to 8:
|                                          |     r[i] = 0;
|                                          |== NEXT i:
|                                          | EXIT; /continue with
|                                          |        section 4.2.6/
|     r[n] = div( abs( P[1] ), P[0] );
|     IF ( P[1] > 0 ) THEN r[n] = sub( 0, r[n] );
|
|     IF (  n ==  8 ) THEN EXIT; /continue with
                                  section 4.2-6/
|     Schur recursion.
|     P[0] = add( P[0], mult_r( P[1], r[n] ) );
|==== FOR m=1 to 8-n:
|       P[m] = add( P[m+1], mult_r( K[9-m], r[n] ) );
|       K[9-m] = add( K[9-m], mult_r( P[m+1], r[n] ) );
|==== NEXT m:
|
|== NEXT n:
```

NOTE: The following lines gives one correct implementation of the div(num, denum) arithmetic operation. Compute div which is the integer division of num by denum: with denum >= num > 0.

```
L_num = num;

L_denum = denum;

div =0;

|== FOR k = 0 to 14:

|     div= div << 1;

|     L_num = L_num << 1;

|     IF ( L_num >= L_denum) THEN

|                               | L_num=L_sub(L_num, L_denum);

|                               | div = add( div ,1 );

|== NEXT k:
```

4.2.6. Transformation of reflection coefficients to Log.-Area Ratios

------

The following scaling for r[..] and LAR[..] has been used:

```
/* r[..] = integer( real_r[..]*32768. );  -1. <= real_r <1. */
/*                                                           */
/* LAR[..] = integer( real_LAR[..]*16384. );                 */
/*                                                           */
/* with  -1.625 <= real_LAR <= 1.625                         */
```

Computation of the LAR[1..8] from the r[1..8].

```
|== FOR i = 1 to 8:
|     temp = abs( r[i] );
|     IF ( temp < 22118 ) THEN temp = temp >> 1;
|        ELSE  IF ( temp < 31130 ) THEN
|               temp= sub(temp, 11059);
|           ELSE temp = sub( temp, 26112 ) << 2;
|     LAR[i] = temp;
|     IF ( r[i] < 0 ) THEN LAR[i] = sub( 0, LAR[i] );
|== NEXT i:
```

4.2.7. Quantization and coding of the Log.-Area Ratios

This procedure needs four tables; the following equations give the optimum scaling for the constants:

```
/* A[1..8]= integer( real_A[1..8]*1024); 8 values (see table4.1)*/
/*                                                               */
/* B[1..8]= integer( real_B[1..8]*512);  8 values (see table4.1)*/
/*                                                               */
/* MAC[1..8]= maximum of the LARc[1..8]; 8 values (see table4.1)*/
/*                                                               */
/* MIC[1..8]= minimum of the LARc[1..8]; 8 values (see table4.1)*/
```

Computation for quantizing and coding the LAR[1..8].

```
|== FOR i =1 to 8:
|     temp= mult( A[i], LAR[i] );
|     temp= add( temp, B[i] );
|     temp= add( temp, 256);     for rounding
|     LARc[i]= temp >> 9;
|   Check IF LARc[i] lies between MIN and MAX
|      IF ( LARc[i] > MAC[i] ) THEN LARc[i] = MAC[i];
|      IF ( LARc[i] < MIC[i] ) THEN LARc[i] = MIC[i];
|     LARc[i] = sub( LARc[i], MIC[i] );  /See note below/
|== NEXT i:
```

NOTE: The equation is used to make all the LARc[i] positive.

SHORT TERM ANALYSIS FILTERING SECTION

4.2.8. Decoding of the coded Log.-Area Ratios

This procedure requires for efficient implementation two tables.

```
/* INVA[1..8]=integer((32768*8)/(real_A[1..8]);                  */
/*                                        8 values (table 4.2 ) */
/* MIC[1..8]=minimum value of the LARc[1..8];                    */
/*                                        8 values (table 4.1)   */
```

Compute the LARpp[1..8].

```
|== FOR i=1 to 8:
|     temp1 = add( LARc[i], MIC[i] ) << 10; /See note below/
|     temp2 = B[i] << 1;
|     temp1 = sub( temp1, temp2);
|     temp1 = mult_r( INVA[i], temp1);
|     LARpp[i] = add( temp1, temp1);
|== NEXT i:
```

NOTE: The addition of MIC[i] is used to restore the sign of LARc[i].

4.2.9. Computation of the quantized reflection coefficients

Within each frame of 160 analysed speech samples the short term analysis and synthesis filters operate with four different sets of coefficients, derived from the previous set of decoded LARs(LARpp(j-1)) and the actual set of decoded LARs (LARpp(j)).

4.2.9.1. Interpolation of the LARpp[1..8] to get the LARp[1..8]

For k_start = 0 to k_end = 12.

```
|==== FOR i= 1 to 8:
|      LARp[i] = add((LARpp(j-1)[i] >> 2),(LARpp(j)[i] >> 2));
|      LARp[i] = add( LARp[i] , ( LARpp(j-1)[i]  >> 1 ) );
|==== NEXT i:
```

For k_start = 13 to k_end = 26.

```
|==== FOR i= 1 to 8:
|      LARp[i] = add((LARpp(j-1)[i] >> 1),(LARpp(j)[i] >> 1 ));
|==== NEXT i:
```

```
For k_start = 27 to k_end = 39.

|==== FOR i= 1 to 8:
|     LARp[i] = add((LARpp(j-1)[i] >> 2),(LARpp(j)[i] >> 2 ));
|     LARp[i] = add( LARp[i] , ( LARpp(j)[i]  >> 1 ) );
|==== NEXT i:


For k_start = 40 to k_end = 159.

|==== FOR i= 1 to 8:
|     LARp[i] =  LARpp(j)[i];
|==== NEXT i:
```

Initial value: LARpp(j-1)[1..8]=0;

4.2.9.2. Computation of the rp[1..8] from the interpolated LARp[1..8]

The input of this procedure is the interpolated LARp[1..8] array. The reflection coefficients, rp[i], are used in the analysis filter and in the synthesis filter.

```
|== FOR i=1 to 8:
|    temp = abs( LARp[i] );
|    IF ( temp < 11059 ) THEN temp = temp << 1;
|        ELSE IF (temp < 20070) THEN
|                 temp = add(temp, 11059);
|           ELSE temp = add( (temp >> 2), 26112 );
|    rp[i] = temp;
|    IF ( LARp[i] < 0 ) THEN rp[i] = sub( 0, rp[i] );
|== NEXT i:
```

4.2.10. Short term analysis filtering

This procedure computes the short term residual signal d[..] to be fed to the RPE-LTP loop from the s[..] signal and from the local rp[..] array (quantized reflection coefficients). As the call of this procedure can be done in many ways (see the interpolation of the LAR coefficient), it is assumed that the computation begins with index k_start (for arrays d[..] and s[..]) and stops with index k_end (k_start and k_end are defined in 4.2.9.1). This procedure also needs to keep the array u[0..7] in memory for each call.

```
|== FOR k = k_start to k_end:
|     di = s[k]
|     sav = di;
|==== FOR i = 1 to 8:
|       temp = add( u[i-1], mult_r( rp[i], di ) );
|       di = add( di, mult_r( rp[i], u[i-1] ) );
|       u[i-1] = sav;
|       sav = temp;
|==== NEXT i:
|     d[k] = di;
|== NEXT k:
```

Keep the array u[0..7] in memory.

Initial value: u[0..7]=0;

LONG TERM PREDICTOR (LTP) SECTION

4.2.11. Calculation of the LTP parameters

This procedure computes the LTP gain (bc) and the LTP lag (Nc) for the long term analysis filter. This is done by calculating a maximum of the cross-correlation function between the current sub-segment short term residual signal d[0..39] (output of the short term analysis filter; for simplification the index of this array begins at 0 and ends at 39 for each sub-segment of the RPE-LTP analysis) and the previous reconstructed short term residual signal dp[-120..-1]. A dynamic scaling must be performed to avoid overflow.

Search of the optimum scaling of d[0..39].

```
dmax = 0;
|== FOR k = 0 to 39:
|     temp = abs( d[k] );
|     IF ( temp > dmax ) THEN dmax = temp;
|== NEXT k:

temp = 0;
IF ( dmax == 0 ) THEN scal = 0;
  ELSE temp = norm( dmax << 16 );
IF ( temp > 6 ) THEN scal = 0;
  ELSE scal = sub( 6, temp );
```

Initialisation of a working array wt[0..39].

```
|== FOR k = 0 to 39:
|     wt[k] = d[k] >> scal;
|== NEXT k:
```

<u>Search for the maximum cross-correlation and coding of the LTP lag</u>.

```
L_max = 0;
Nc = 40;   (index for the maximum cross-correlation)

|== FOR lambda = 40 to 120:
|    L_result = 0;
|==== FOR k = 0 to 39:
|      L_temp = L_mult( wt[k], dp[k-lambda] );
|      L_result = L_add( L_temp, L_result );
|==== NEXT k:
|    IF ( L_result > L_max) THEN
|                                  | Nc = lambda;
|                                  | L_max = L_result ;
|== NEXT lambda:
```

<u>Rescaling of L-max</u>.

```
L_max = L_max >> ( sub( 6, scal ) );
```

<u>Initialisation of a working array wt[0..39]</u>.

```
|== FOR k = 0 to 39:
|    wt[k] = dp[k-Nc] >> 3;
|== NEXT k:
```

Compute the power of the reconstructed short term residual signal dp[..].

```
L_power = 0;

|== FOR k =0 to 39:

|     L_temp = L_mult( wt[k], wt[k] );

|     L_power = L_add( L_temp, L_power );

|== NEXT k:
```

Normalization of L-max and L-power.

```
IF ( L_max <= 0 ) THEN
                              | bc = 0;
                              | EXIT; /cont. with 4.2-12/
IF ( L_max >= L_power ) THEN
                                    | bc = 3;
                                    | EXIT; /cont. with 4.2-12/
temp = norm( L_power );

R = ( L_max << temp ) >> 16;

S = ( L_power << temp ) >> 16;
```

Coding of the LTP gain.

Table 4.3a must be used to obtain the level DLB[i] for the quantization of the LTP gain b to get the coded version bc.

```
|== FOR bc = 0 to 2:

|      IF (R <= mult(S, DLB[bc])) THEN EXIT; /cont. with
                                                4.2.12/
|== NEXT bc;

bc = 3;
```

Initial value: dp[-120..-1]=0;

4.2.12. Long term analysis filtering

In this part, we have to decode the bc parameter to compute the samples of the estimate dpp[0..39]. The decoding of bc needs the use of table 4.3b. The long term residual signal e[0..39] is then calculated to be fed to the RPE encoding section.

Decoding of the coded LTP gain.

```
bp = QLB[bc];
```

Calculating the array e[0..39] and the array dpp[0..39].

```
|== FOR k = 0 to 39:
|     dpp[k] = mult_r( bp, dp[k-Nc] );
|     e[k] = sub( d[k], dpp[k] );
|== NEXT k:
```

RPE ENCODING SECTION

4.2.13. Weighting filter

The coefficients of the weighting filter are stored in a table (see table 4.4). The following scaling is used:

```
/* H[0..10] = integer( real_H[0..10]*8192 ); */
```

Initialisation of a temporary working array wt[0..49].

```
|== FOR k= 0 to 4:
|     wt[k] = 0;
|== NEXT k:
```

```
|== FOR k = 5 to 44:
|    wt[k] = e[k-5];
|== NEXT k:

|== FOR k= 45 to 49:
|    wt[k] = 0;
|== NEXT k:
```

Compute the signal x[0..39].

```
|== FOR k= 0 to 39:
|      L_result =  8192; /rounding of the output
                                  of the filter/
|==== FOR i = 0 to 10:
|      L_temp = L_mult( wt[k+i], H[i] );
|      L_result = L_add( L_result, L_temp );
|==== NEXT i:
|      L_result = L_add(L_result,L_result); /scaling (x2)/
|      L_result = L_add(L_result,L_result); /scaling (x4)/
|    x[k] = L_result >> 16;
|== NEXT k:
```

4.2.14. RPE grid selection

The signal x[0..39] is used to select the RPE grid which is represented by Mc.

```
EM =0;

Mc = 0;


|== FOR m = 0 to 3:

|    L_result = 0;

|==== FOR i = 0 to 12:

|       temp1 = x[m+(3*i)] >> 2;

|       L_temp = L_mult( temp1, temp1 );

|       L_result = L_add( L_temp, L_result );

|==== NEXT i:

|    IF ( L_result > EM) THEN

|                                      | Mc = m;

|                                      | EM = L_result;

|== NEXT m:
```

Down-sampling by a factor 3 to get the selected xM[0..12] RPE sequence.

```
|== FOR i = 0 to 12:

|    xM[i] = x[Mc +(3*i)];

|== NEXT i:
```

4.2.15. APCM quantization of the selected RPE sequence

Find the maximum absolute value xmax of xM[0..12].

```
xmax = 0;
|== FOR i = 0 to 12:
|     temp = abs( xM[i] ) ;
|     IF ( temp > xmax ) THEN xmax = temp;
|== NEXT i:
```

Quantizing and coding of xmax to get xmaxc.

```
exp = 0;
temp = xmax >> 9;
itest = 0;
|== FOR i = 0 to 5:
|     IF ( temp <= 0 ) THEN itest = 1;
|     temp = temp >> 1;
|     IF ( itest == 0 ) THEN exp = add( exp, 1 ) ;
|== NEXT i:

temp = add( exp, 5 ) ;
xmaxc = add( ( xmax >> temp ), ( exp << 3 ) ) ;
```

Quantizing and coding of the xM[0..12] RPE sequence to get the xMc[0..12].

This computation uses the fact that the decoded version of xmaxc can be calculated by using the exponent and the mantissa part of xmaxc (logarithmic table).

So, this method avoids any division and uses only a scaling of the RPE samples by a function of the exponent. A direct multiplication by the inverse of the mantissa (NRFAC[0..7] found in table 4.5) gives the 3 bit coded version xMc[0..12] of the RPE samples.

Compute exponent and mantissa of the decoded version of xmaxc.

```
exp = 0 ;
IF ( xmaxc > 15 ) THEN exp = sub( ( xmaxc >> 3 ), 1 ) ;
mant = sub( xmaxc , ( exp << 3 ) );
```

Normalize mantissa 0 <= mant <= 7.

```
IF ( mant == 0 ) THEN | exp = -4;
                      | mant = 15;
ELSE | itest = 0;
     |== FOR i = 0 to 2:
     |     IF ( mant > 7 ) THEN itest = 1;
     |     IF (itest == 0) THEN mant = add((mant << 1),1);
     |     IF ( itest == 0 ) THEN exp = sub( exp, 1 );
     |== NEXT i:

mant = sub( mant, 8 );
```

Direct computation of xMc[0..12] using table 4.5.

```
temp1= sub( 6, exp );  /normalization by the exponent/
temp2  =  NRFAC[mant];  /see table 4.5 (inverse mantissa)/

|== FOR i = 0 to 12:
|    temp = xM[i] << temp1;
|    temp = mult( temp , temp2 );
|    xMc[i] = add( ( temp >> 12 ), 4 );  /See note below/
|== NEXT i:
```

NOTE: This equation is used to make all the xMc[i] positive.

Keep in memory exp and mant for the following inverse APCM quantizer.

4.2.16. APCM inverse quantization

This part is for decoding the RPE sequence of coded xMc[0..12] samples to obtain the xMp[0..12] array. Table 4.6 is used to get the mantissa of xmaxc (FAC[0..7]).

```
temp1 = FAC[mant];  see 4.2-15 for mant
temp2= sub( 6, exp );  see 4.2-15 for exp
temp3= 1 << sub( temp2, 1 );
```

```
|== FOR i =0 to 12:
|     temp = sub( ( xMc[i] << 1 ), 7 );  /See note below/
|     temp = temp << 12;
|     temp = mult_r( temp1, temp );
|     temp = add( temp, temp3 );
|     xMp[i] = temp >> temp2;
|== NEXT i;
```

NOTE: This subtraction is used to restore the sign of xMc[i].

4.2.17. RPE grid positioning

This procedure computes the reconstructed long term residual signal ep[0..39] for the LTP analysis filter. The inputs are the Mc which is the grid position selection and the xMp[0..12] decoded RPE samples which are upsampled by a factor of 3 by inserting zero values.

```
|== FOR k = 0 to 39:
|     ep[k] = 0;
|== NEXT k:


|== FOR i = 0 to 12:
|     ep[Mc +(3*i)] = xMp[i];
|== NEXT i:
```

4.2.18. Update of the reconstructed short term residual signal dp[-120..-1]

This procedure adds the reconstructed long term residual signal ep[0..39] to the estimated signal dpp[0..39] from the long term analysis filter to compute the recontsructed short term residual signal dp[-40..-1]; also the reconstructed short term residual array dp[-120..-41] is updated.

```
|== FOR k = 0 to 79:
|    dp[-120+k] = dp[-80+k];
|== NEXT k:

|== FOR k = 0 to 39:
|    dp[-40+k] = add( ep[k], dpp[k] );
|== NEXT k:
```

Keep the array dp[-120..-1] in memory for the next sub-segment.

Initial value: dp[-120..-1]=0;

4.3. FIXED POINT IMPLEMENTATION OF THE RPE-LTP DECODER

Only the synthesis filter and the de-emphasis procedure are different from the procedures found in the RPE-LTP coder. Procedures 4.3.1 and 4.3.2 are executed for each sub-segment (four times per frame). Procedures 4.3.3, 4.3.4 and 4.3.5 are executed once per frame.

4.3.1. RPE decoding section

Procedures 4.2.15 (only the part to get mant and exp of xmaxc), 4.2.16 and 4.2.17 are used to obtain the reconstructed long term residual signal erp[0..39] signal from the received parameters for each sub-segment (ie Mcr, xmaxcr, xmcr[0..12]).

4.3.2. Long term synthesis filtering

This procedure uses the bcr and Ncr parameter to realize the long term synthesis filtering. The decoding of bcr needs the use of table 4.3b.

- Nr is the received and decoded LTP lag.
- An array drp[-120..39] is used in this procedure.

The elements for -120 to -1 of the array drp are kept in memory for the long term synthesis filter. For each sub-segment (40 samples), this procedure computes the drp[0..39] to be fed to the synthesis filter.

Check the limits of Nr.

```
Nr = Ncr;

IF ( Ncr < 40 ) THEN Nr = nrp;

IF ( Ncr > 120 ) THEN Nr = nrp;

nrp= Nr;
```

Keep the nrp value for the next sub-segment.

Initial value: nrp=40;

Decoding of the LTP gain bcr.

```
brp = QLB[bcr]
```

Computation of the reconstructed short term residual signal drp[0..39].

```
|== FOR k = 0 to 39:
|     drpp = mult_r( brp, drp[k-Nr] );
|     drp[k] = add( erp[k], drpp );
|== NEXT k:
```

Update of the reconstructed short term residual signal drp[-1..-120].

```
|== FOR k = 0 to 119:
|     drp[-120+k] = drp[-80+k];
|== NEXT k:
```

Keep the array drp[-120..-1] for the next sub-segment.

Initial value: drp[-120..-1]=0;

4.3.3. Computation of the decoded reflection coefficients

This procedure (which is executed once per frame) is the same as the one described in the CODER part.

For decoding of the received LARcr[1..8], see procedure 4.2.8.

For the interpolation of the decoded Log.-Area Ratios, see procedure 4.2.9.1 and for the computation of the reflection coefficients rrp[1..8], see procedure 4.2.9.2.

4.3.4. Short term synthesis filtering section

This procedure uses the drp[0..39] signal and produces the sr[0..159] signal which is the output of the short term synthesis filter. For ease of explanation, a temporary array wt[0..159] is used.

Initialisation of the array wt[0..159].

For the first sub-segment in a frame:

```
|== FOR k = 0 to 39:
|     wt[k] = drp[k];
|== NEXT k:
```

For the second sub-segment in a frame:

```
|== FOR k = 0 to 39:
|     wt[40+k] = drp[k];
|== NEXT k:
```

For the third sub-segment in a frame:

```
|== FOR k = 0 to 39:
|     wt[80+k] = drp[k];
|== NEXT k:
```

For the fourth sub-segment in a frame:

```
|== FOR k = 0 to 39:
|     wt[120+k] = drp[k];
|== NEXT k:
```

As the call of the short term synthesis filter procedure can be done in many ways (see the interpolation of the LAR coefficient), it is assumed that the computation begins with index k_start (for arrays wt[..] and sr[..]) and stops with index k_end (k_start and k_end are defined in 4.2.9.1). The procedure also needs to keep the array v[0..8] in memory between calls.

```
|== FOR k = k_start to k_end:
|    sri = wt[k];
|==== FOR i = 1 to 8:
|      sri = sub( sri, mult_r( rrp[9-i], v[8-i] ) );
|      v[9-i] = add( v[8-i], mult_r( rrp[9-i], sri ) );
|==== NEXT i:
|    sr[k] = sri;
|    v[0] = sri;
|== NEXT k:
```

<u>Keep the array v[0..8] in memory for the next call</u>.

<u>Initial value</u>: v[0..8]=0;

<u>POSTPROCESSING</u>

4.3.5. Deemphasis filtering

```
|== FOR k = 0 to 159:
|    temp = add( sr[k], mult_r( msr, 28180 ) );
|    msr = temp;
|    sro[k] = msr;
|== NEXT k:
```

<u>Keep msr in memory for the next frame</u>.

<u>Initial value</u>: msr=0;

4.3.6. Upscaling of the output signal

```
|== FOR k = 0 to 159:
|    srop[k] = add( sro[k], sro[k] );
|== NEXT k:
```

4.3.7. Truncation of the output variable

```
|== FOR k = 0 to 159:
|    srop[k] = srop[k] >> 3;
|    srop[k] = srop[k] << 3;
|== NEXT k:
```

The output format is the following:

S.v.v.v.v.v.v.v.v.v.v.v.v.0.0.0 (2's complement).

Where S is the sign bit, v a valid bit.

<u>NOTE</u>: When a linear to A-law compression is needed, then the sub-block COMPRESS of CCITT G721 recommendation must be used with inputs:

```
SR = srop[k] >> 3;
LAW = 1;
```

4.4. TABLES USED IN THE FIXED POINT IMPLEMENTATION OF THE RPE-LTP CODER AND DECODER

i	A[i]	B[i]	MIC[i]	MAC[i]
1	20480	0	-32	31
2	20480	0	-32	31
3	20480	2048	-16	15
4	20480	-2560	-16	15
5	13964	94	-8	7
6	15360	-1792	-8	7
7	8534	-341	-4	3
8	9036	-1144	-4	3

Table 4.1. Quantization of the Log.-Area Ratios

i	INVA[i]
1	13107
2	13107
3	13107
4	13107
5	19223
6	17476
7	31454
8	29708

Table 4.2. Tabulation of 1/A[1..8]

bc	DLB[bc]
0	6554
1	16384
2	26214
3	32767

Table 4.3a. Decision level of the LTP gain quantizer

bc	QLB[bc]
0	3277
1	11469
2	21299
3	32767

Table 4.3b. Quantization levels of the LTP gain quantizer

i	H[i]
0	-134
1	-374
2	0
3	2054
4	5741
5	8192
6	5741
7	2054
8	0
9	-374
10	-134

Table 4.4. Coefficients of the weighting filter

i	NRFAC[i]
0	29128
1	26215
2	23832
3	21846
4	20165
5	18725
6	17476
7	16384

Table 4.5. Normalized inverse mantissa used to compute xM/xmax

i	FAC[i]
0	18431
1	20479
2	22527
3	24575
4	26623
5	28671
6	30719
7	32767

Table 4.6. Normalized direct mantissa used to compute xM/xmax

5. DIGITAL TEST SEQUENCES

This chapter provides information on the digital test sequences that have been designed to help in the verification of implementations of the RPE-LTP codec. Copies of these sequences are available (see annex 3).

5.1. INPUT AND OUTPUT SIGNALS

Table 5.1 defines the input and output signals for the test sequences. The words defined in this table use 16 bits. The left or right justification is indicated in the table.

The codewords described in the table correspond to one frame of coder input or decoder output signal; i.e for 20 ms of input signal the 76 codewords are obtained at the output of the coder and 76 codewords provided at the input of the decoder will yield 20 ms of output signal in the decoder.

5.2. CONFIGURATION FOR THE APPLICATION OF THE TEST SEQUENCES

Two configurations are appropriate in order to test an implementation of the RPE-LTP coder. The first one is for testing the coder part of the RPE-LTP; it means that a sop[..] signal is provided at the input of the encoder that furnishes frames of coded parameters. This output has to be checked against some reference file. The other configuration is for testing the decoder part of the RPE-LTP; in this case frames of coded parameters (see table 5.1) are sent to the RPE-LTP decoder that furnishes the srop[..] signal. These samples have to be checked against a reference file.

5.2.1. Configuration 1 (encoder only)

A reset signal (RS) must be applied to the RPE-LTP encoder under test to set all internal variables to the exact states specified in section 4 of this recommendation prior to the start of an input test sequence in order to obtain the correct output values for this test. This test must be done in real time with a sampling rate of 8 kHz at the input of the encoder under test (see figure 5.1). All the necessary hardware and software should be installed by the user in order to capture in real time the output coded parameters of the RPE-LTP encoder and to compare them to the dedicated reference file.

```
                      | 8 kHz sampling rate
                      v
                   --------------
        RS --->| RPE-LTP      |  Coded parameters
               | Encoder      |------------/-------->  Comparison
Input:   ----/->| under test  |            2-7 bits
       13 bits  --------------
```

Figure 5.1. Configuration 1: RPE-LTP encoder under test

5.2.2. Configuration 2 (Decoder only)

Figure 5.2 shows a RPE-LTP decoder under test. In the same way as described in the coder part, a reset signal (RS) must be used before the processing of the first frame of coded parameters. The decoder must be tested for a continuous output with a sampling rate of 8 kHz. At the input of the decoder, the 76 parameters must be sent in a time interval of 20 ms.

```
                          | 8 kHz sampling rate
                          v
                       ------------
          RS ---->| RPE-LTP    |  Output signal: srop[k]
                  | Decoder    |-----------/-------> Comparison
Input: coded ---/-->| under test |          13 bits
     parameters        ------------
     2-7 bits
```

Figure 5.2. Configuration 2: RPE-LTP decoder under test

5.3. TEST SEQUENCES

5.3.1. Test sequences for configuration 1

For configuration 1, four types of input test sequence are provided:

1. Sequence for testing the overflow controls in the encoder
2. Sequence for testing the LPC part of the encoder
3. Sequence for testing the LTP part of the encoder
4. Sequence for testing various critical parts of the algorithm

Sequence 1 uses a large number of saturated samples. The residual LPC signal reaches very high values, which has two effects on the processing:

- occurrence of a large number of overflows in addition/subtraction operations. Table 5.2 describes each overflow point and the number of occurrences for each.

- the excitation RPE samples have a large dynamic range and the 64 codewords of the sub-block maximum are each obtained at least once on output.

Sequence 2 focuses successively on each reflection coefficient calculated in the Schur recursion. Table 5.3 shows which frames deal with which reflection coefficient and its dynamic range. The Log.-Area codewords output by the coder cover the full range of their possible values except the 2nd LARc that does not reach the value 0 and 63 (min and max). The maximum value (63) is however obtained in sequence 4.

Sequence 3 tests the long term predictor part of the algorithm. It has been generated by exciting a sharply resonant filter with a periodic train of impulses; this produces a pitched signal. Each part corresponding to a given pitch is 128 ms (4 blocks of 256 words) long. The pitch periods have been randomly drawn in the range [2,15] ms and the random order is shown in table 5.4.

Sequence 4 accounts for various remaining non tested points of the algorithm where implementing errors may be suspected. Table 5.5 and 5.6 summarize the critical points that this sequence has been designed to check (ie where the three previous sequences were ineffective). Table 5.5 shows the list of tested points where errors can be detected. Each tested error is described and the frame number corresponding to the first occurrence of a divergence between the exact and the degraded algorithm is also indicated.

Table 5.6 illustrates three paths of the algorithm that are never explored during the processing of the three previous sequences; the table shows which condition leads to each path and the number of associated occurrences in sequence 4.

Notice finally one point where special care must be taken:

- A small degradation (ie +/- 1) of DLB[2] (the third decision level of the LTP gain quantizer (see table 4.3a) is unable to provide any noticeable effect on the output of the four sequences described above).

5.3.2. Test sequences for configuration 2

Five types of input test sequence are provided for this configuration. Four sequences obtained in configuration 1 at the output of the encoder (coded parameters) are used as input for the decoder under test in configuration 2.

Table 5.7 gives the list of tested overflow points and their occurrence on sequence 1 for this configuration.

Sequence 5 is provided to scan all possible codes for each parameter. This sequence is an artificial sequence and does not correspond to any encoder output. The codewords have been randomly generated and cover the entire range of codewords values. Moreover, the delay value Nr belonging to [40,120] in an error-free transmission condition, takes in this sequence its value in [0,127]. In this case the decoder behaviour on non-allowed values of Nr will be tested.

Name	Description	Justification
	ENCODER INPUT	
sop[k]	13 bits: encoder input signal.	left
	OUTPUT PARAMETERS	
LARc[1]	6 bits : 1st Log.-Area Ratio	right
LARc[2]	6 bits : 2nd Log.-Area Ratio	right
LARc[3]	5 bits : 3rd Log.-Area Ratio	right
LARc[4]	5 bits : 4th Log.-Area Ratio	right
LARc[5]	4 bits : 5th Log.-Area Ratio	right
LARc[6]	4 bits : 6th Log.-Area Ratio	right
LARc[7]	3 bits : 7th Log.-Area Ratio	right
LARc[8]	3 bits : 8th Log.-Area Ratio	right
	Sub-frame no 1	
Nc	7 bits : LTP lag	right
bc	2 bits : LTP gain	right
Mc	2 bits : RPE grid position	right
xmaxc	6 bits : Block amplitude	right
xMc[0..12]	3 bits : RPE pulses index 0 to 12	right
	Sub-frame no 2	
Nc	7 bits : LTP lag	right
bc	2 bits : LTP gain	right
Mc	2 bits : RPE grid position	right
xmaxc	6 bits : Block amplitude	right
xMc[0..12]	3 bits : RPE pulses index 0 to 12	right
	Sub-frame no 3	
Nc	7 bits : LTP lag	right
bc	2 bits : LTP gain	right
Mc	2 bits : RPE grid position	right
xmaxc	6 bits : Block amplitude	right
xMc[0..12]	3 bits : RPE pulses index 0 to 12	right
	Sub-frame no 4	
Nc	7 bits : LTP lag	right
bc	2 bits : LTP gain	right
Mc	2 bits : RPE grid position	right
xmaxc	6 bits : Block amplitude	right
xMc[0..12]	3 bits : RPE pulses index 0 to 12	right

Table 5.1a. Signals used in digital test sequences

DECODER OUTPUT		
srop[k]	13 bits: decoder output signal. The 3 LSB's of the 16 bits are equal to 0.	left

Table 5.1b. Signals used in digital test sequences

Overflow point	No of occurrences
Short term analysis filter (4.2.10)	
1st add	1059
2nd add	134
LTP parameters computation (4.2.11)	
Abs(d[k])	5
Long term analysis filter (4.2.12)	
sub	11
Weighting filter (4.2.13)	
scaling the result (both x2 and x4)	302
APCM quantizer (4.2.15)	
Find max abs of xm: Abs	49
Update of Array dp of the long term analysis filter (4.2.18)	
add	126

Table 5.2. List of tested overflow points for sequence 1 (coder part)

Reflection Coeff.	Frames	Dynamic range
1	1-135	-32564,32558
2	136-311	-32356,32242
3	316-423	-32157,32744
4	424-524	-31594,31960
5	525-633	-31697,31735
6	634-738	-30055,31575
7	739-839	-29090,31386
8	840-944	-31052,31208

Table 5.3.

Table 5.3 gives the position of the frames dedicated to the study of each reflection coefficient and dynamic range of the coefficient for sequence 2 in configuration 2.

86	56	68	120	52	93	20	66	82	115	114	60	42	45	17	64	16
88	83	63	90	73	23	77	100	33	29	106	35	67	57	103	116	30
71	69	81	47	32	97	65	62	111	49	109	25	96	50	54	91	85
99	70	76	46	26	34	104	108	107	22	119	48	58	37	72	110	27
24	36	87	51	59	38	21	44	113	39	61	53	18	40	94	105	55
112	75	98	118	41	80	31	74	28	84	89	79	43	101	95	19	78
117	92	102														

Table 5.4. Pitch periods of sequence 3 (configuration 1)

Test point	Error checked : incorrect statement / correct statement	No of the 1st frame with error
Autocorrelation function (4.2.4)	k=0 to 158 / k=0 to 159	27
Computation of the reflect. coefficients (4.2.5)	if(P[0] <= abs(P[1])) / if(P[0] < abs(P[1]))	514
Quantization and coding of the LARs (4.2.7)	A[4] + 1 / A[4]	21
	A[5] - 1 / A[5]	35
	A[5] + 1 / A[5]	430
	A[6] - 1 / A[6]	427
	A[8] - 1 / A[8]	8
	MAC[2] - 1 / MAC[2]	24
	MAC[2] + 1 / MAC[2]	516
Comput. of the rp from the interp. LARp (4.2.9)	11058 / 11059	19
	20069 / 20070	25
Calc. of the LTP parameters ->Search of the opt scaling (4.2.11)	k= 0 to 38 / k= 0 to 39	32
->Coding of the LTP gain (4.2.11)	mult_r / mult	373
	DLB[0] + 1 / DLB[0]	511
	DLB[1] + 1 / DLB[1]	373
ADPCM inverse quantizer (4.2.16)	FAC[2] + 1 / FAC[2]	422
	FAC[3] - 1 / FAC[3]	179
	FAC[4] + 1 / FAC[4]	74
	FAC[5] - 1 / FAC[5]	439
	FAC[5] + 1 / FAC[5]	74
	FAC[6] - 1 / FAC[6]	479
	FAC[6] + 1 / FAC[6]	330
	FAC[7] - 1 / FAC[7]	139

Table 5.5. Errors specially detected by sequence 4/Config 1

Test point	Number of occurrences
Autocorrelation function (4.2.4) condition smax == 0	8
Computation of the reflection coefficients :	
-> condition L_ACF[0] == 0 (4.2.5)	8
-> condition P[0] < abs(P[1]) (4.2.5)	4

Table 5.6. Paths specially explored by sequence 4/Config 1

Overflow Point	Nb of occurrences Sequence 1	Sequence 2	Sequence 3
Long term synthesis filter (4.3.2) : add	126	0	0
Short term synthesis filter (4.3.4) :			
1st add:	4499	0	0
2nd add:	405	1	0
De-emphasize filter (4.3.5) : add	89	0	0
Scaling of the output signal (4.3.6) : add	16691	339	19

Table 5.7. List of tested overflows points for sequence 1 (decoder part)

ANNEX 1. CODEC PERFORMANCE

A1.1. INTRODUCTION

The purpose of this annex is to give a broad outline of the performance of the RPE-LTP codec with other parts of the digital network. Some general guidance is also offered on non-voice services.

A1.2. SPEECH PERFORMANCE

Planning rules for digital processes are defined in terms of quantizing distortion units (qdu) which can be realized from the following formula (reference 1) using the assumption that the formula accuracy represents the determination of qdus from QN measurements:

$$QN = 37 - 15 \log_{10}(n) \quad \text{,where n is the qdu} \qquad \text{(A1.1)}$$

By definition 1 qdu is the quantization distortion arising from one commercial PCM codec.

NOTE: The subjective testing methodology to determine QN for the RPE-LTP codec was consistent with current CCITT methods (reference 2).

A1.2.1. Single encoding

Under error-free transmission conditions the perceived quality of the RPE-LTP codec (see fig A1.1) is lower than both codecs conforming to recommendations CCITT G.711 and CCITT G.721.

Table A1.1 indicates the relative performance of the codec and can be compared with codecs conforming to recommendations CCITT G.711 and CCITT G.721.

The performance of the RPE-LTP codec has been found to be substantially unaffected down to a carrier to interference (C/I) ratio of 10 dB, but may be considered to have acceptable performance down to 7 dB. Smaller C/I ratios produce unacceptable degradation of speech performance and should be avoided.

NOTE 1: It should be noted that there are doubts as to whether the simulations which generated the error pattern properly represent real operating conditions. The C/I values quoted should therefore only be considered as parameters of this simulation. They may not correspond to real radio interference conditions. Results from early GSM validation hardware show that the C/I values which give the performance quoted may be several dBs higher. Some error statistics of the simulations are shown in table A1.2.

NOTE 2: The real condition C/I = 10 dB is believed to correspond to about 90 % coverage.

Codec	QN (dB)	qdu
G.711 (64 kbit/s, A-law PCM)	37	1
G.721 (32 kbit/s, ADPCM)	29	3.5 (*)
RPE-LTP	23-25	7-8 (*)

(*) Commercial A-law PCM input and output circuitry included.

NOTE: The qdu value for the RPE-LTP codec is a conservative estimate. At present there are no specific CCITT rules for determining qdus for encoding below 32 kbit/s.

Table A1.1. Relative levels of speech performance under error-free conditions

Simulated C/I ratio:	10 dB	7 dB	4 dB
Total number of errors in class I (182 bits protected by a 1/2 rate code)	0.016%	0.61%	4.1%
Total number of errors in class II (78 bits unprotected)	4.5%	8.3%	13.0%
Number of "frame erasure" indications by CRC	1	15	95
Number of "frame erasures" not detected by CRC	1	14	76

NOTE: The total number of frames was 750. CRC means Cyclic Redundancy Check.

Table A1.2. Bit error statistics for C/I test conditions

A1.2.2. Speech performance when interconnected with coding systems on an analogue basis

A1.2.2.1. Performance with 32 kbit/s ADPCM (G.721)

The speech performance of the RPE-LTP codec when interconnected with encoding at 32 kbit/s (see fig A1.3 and A1.4) decreases in accordance with the formula in section A1.2, and appears to obey the law of additivity when qdus have been determined for the individual codecs.

A1.2.2.2. Performance with another RPE-LTP codec

The speech performance of the RPE-LTP codec when interconnected with another codec of the same type (see fig A1.2) is lower than that of A1.2.2.1. It again appears to obey the law of additivity when qdus have been determined for the individual codecs.

A1.2.2.3. Performance with encoding other than RPE-LTP and 32 kbit/s ADPCM (G.721)

No information is available on this point, so great care must be exercised when interconnection is made to codecs with encoding different from that of A1.2.2.1 and A1.2.2.2.

A1.3. NON-SPEECH PERFORMANCE

It should be noted that the RPE-LTP speech codec is an adaptive system which has been optimised for speech inputs. Great care must be taken when making measurements with non-speech signals beause the normal assumptions of time invariance and linearity cannot be made.

A.1.3.1. Performance with single sine waves

Detailed experiments have shown that the RPE-LTP codec will pass sine waves with segmental signal to noise ratios generally in excess of 20 dB in the frequency range of 100 - 2000 Hz. However, in some cases reproduction above 2000 Hz is not as good.

It should be noted that sine waves above 1300 Hz may be reproduced with significant fluctuations in amplitude and frequency due to the adaptive sub-sampling technique employed. This results in irregularities in the measured frequency response.

A typical frequency response measured with A-law PCM input circuitry is shown in fig A1.5. If 13 bit linear PCM input circuitry is used, the irregularity is less.

A1.3.2. Performance with DTMF tones

It has been shown that the RPE-LTP codec transfers DTMF signals of 80 ms duration. However, questions like minimum allowable signal duration, pause duration and the behaviour in the presence of transmission errors have not been investigated.

A1.3.3. Performance with information tones

Experiments have shown that network originated signalling tones, conforming to recommendation CCITT Q.35 (reference 3), are easily recognizable when passed through the RPE-LTP codec.

A1.3.4. Performance with voice-band data

Tests have shown that voice-band data transmission does not work satisfactorily with 1200 bit/s modems according to recommendation CCITT V.23. Voice-band data according to recommendation CCITT V.21 (300 bit/s) will not be subject to any significant degradation.

This behaviour has been tested for one RPE-LTP link (encoder-decoder). The effect of transmission errors has not been tested.

A1.4. DELAY

The theoretical minimum delay of the RPE-LTP codec is 20 ms. However, practical realizations may have an additional processing time in the order of 3 - 8 ms.

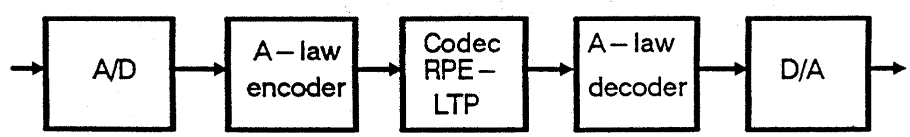


Fig A.1.1. One – transcoding scheme

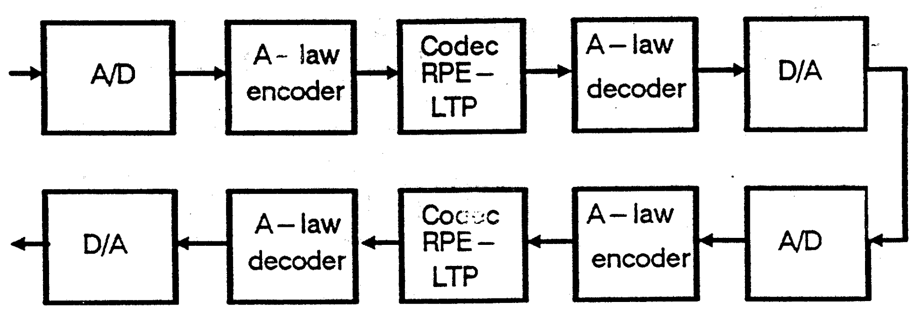


Fig A1.2. Two – transcodings scheme

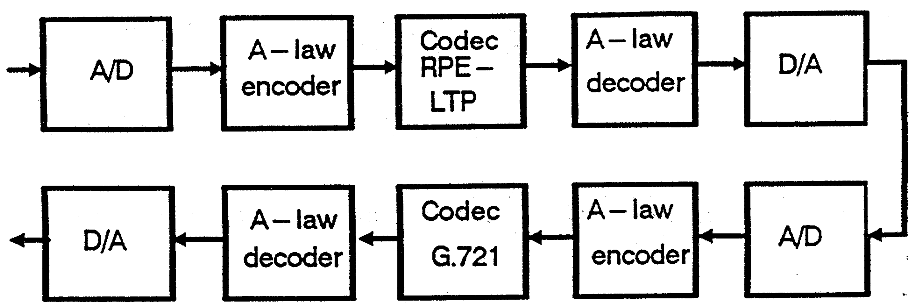


Fig A1.3. Mixed transcodings – scheme 1

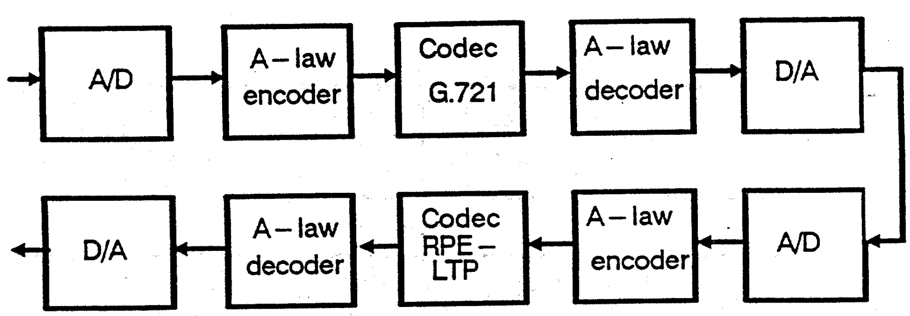


Fig A1.4. Mixed transcodings – scheme 2

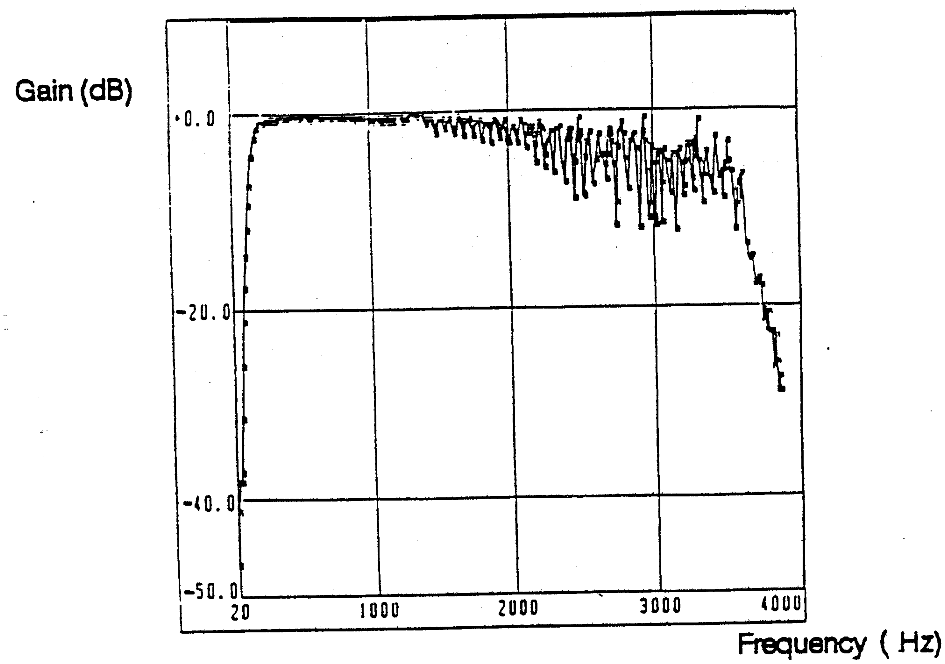


Fig A1.5. Frequency response for RPE-LTP codec (with commercial A-law PCM input and output circuitry)

A1.5. REFERENCES

1. CCITT: "Subjective performance assessment of digital processes using the Modulated Noise Reference Unit (MNRU)", Annex C, Supplement no 14, Red book, volume V, 1985.

2. CCITT: "Subjective performance assessment of digital processes using the Modulated Noise Reference Unit (MNRU)", Annex A, Supplement no 14, Red book, volume V, 1985.

3. CCITT: "Technical characteristics of tones for the telephone service", recommendation Q.35, Red book, volume VI.1, 1985.

ANNEX 2. SUBJECTIVE RELEVANCE OF THE SPEECH CODER OUTPUT BITS

Since no valid objective quality criterion for speech signals is available, the only way to build up such a relevance table is to perform listening tests. The procedure described below was used to obtain the relevance classification given in table A2.1 of the recommendation.

To classify a single bit, say bit i of parameter k, a short speech signal (2 sec) was encoded, then this bit was inverted in each frame (the other bits were left unchanged) and the resulting bit stream was fed into the speech decoder. The listeners had to compare the quality of the signal with the quality of six reference signals with different levels of distortion. Repeating this procedure for all bits would result in a subdivision of the 260 bits into six relevance classes.

It can be observed that many of the bits have the same physical meaning and it can be expected that bits with the same meaning have the same relevance (eg the MSB's of the RPE samples). Relying on this assumption, only one of the equivalent parameters was considered. Since there are 13 parameters with different physical meaning with 56 bits in total, the number of tests is reduced from 260 to 56.

The reference signals were the same speech signal distorted by inverting one of the six bits of LAR coefficient number one. This resulted in an adequate quantization of distortion levels ranging from "not intelligible" (MSB inverted) to "negligible distortion" (LSB inverted).

The test was carried out using three listeners and one female speaker. Since the three listeners came to rather similar results, no more listeners were considered to be required. Averaging the three outcomes led to the relevance table given in table A2.1, where the order of all bits between two successive bits of the first parameter (LAR 1) are arbitrarily chosen.

Importance class	Parameter name	Parameter number	Bit number
1	Log.area ratio 1	1	b6
	Block amplitude	12,29,46,63	b53,b109,b165,b221
2	Log.area ratio 1	1	b5
	Log.area ratio 2	2	b12
	Log.area ratio 3	3	b17
3	Log.area ratio 1	1	b4
	Log.area ratio 2	2	b11
	Log.area ratio 3	3	b16
	Log.area ratio 4	4	b22
	LTP lag	9,26,43,60	b43,b99,b155,b211
	Block amplitude	12,29,46,63	b52,b108,b164,b220
	Log.area ratio 2,5,6	2,5,6	b10,b26,b30
	LTP lag	9,26,43,60	b42,b98,b154,b210
	LTP lag	9,26,43,60	b41,b97,b153,b209
	LTP lag	9,26,43,60	b40,b96,b152,b208
	LTP lag	9,26,43,60	b39,b95,b151,b207
4	Block amplitude	12,29,46,63	b51,b107,b163,b219
	Log.area ratio 1	1	b3
	Log.area ratio 4	4	b21
	Log.area ratio 7	7	b33
	LTP lag	9,26,43,60	b38,b94,b150,b206
	Log.area ratio 5,6	5,6	b25,b29
	LTP gain	10,27,44,61	b45,b101,b157,b213
	LTP lag	9,26,43,60	b37,b93,b149,b205
	Grid position	11,28,45,62	b47,b103,b159,b215

Table A2.1a. Subjective importance of encoded bits (the parameter and bit numbers refer to table 1.1)

Importance class	Parameter name	Parameter number	Bit number
	Log.area ratio 1	1	b2
	Log.area ratio 2,3,8,4	2,3,8,4	b9,b15,b36,b20
	Log.area ratio 5,7	5,7	b24,b32
	LTP gain	10,27,44,61	b44,b100,b156,b212
	Block amplitude	12,29,46,63	b50,b106,b162,b218
	RPE pulses	13..25	b56,b59,..,b92
	RPE pulses	30..42	b112,b115,..,b148
	RPE pulses	47..59	b168,b171,..,b204
5	RPE pulses	64..76	b224,b227,..,b260
	Grid position	11,28,45,62	b46,b102,b158,b214
	Block amplitude	12,29,46,63	b49,b105,b161,b217
	RPE pulses	13..25	b55,b58,..,b91
	RPE pulses	30..42	b111,b114,..,b147
	RPE pulses	47..59	b167,b170,..,b203
	RPE pulses	64..67	b223,b226,b229,b232
	RPE pulses	68..76	b235,b238,..,b259
	Log.area ratio 1	1	b1
	Log.area ratio 2,3,6	2,3,6	b8,b14,b28
	Log.area ratio 7	7	b31
	Log.area ratio 8	8	b35
	Log.area ratio 8,3	8,3	b34,b13
	Log.area ratio 4	4	b19
6	Log.area ratio 4,5	4,5	b18,b23
	Block amplitude	12,29,46,63	b48,b104,b160,b216
	RPE pulses	13..25	b54,b57,..,b90
	RPE pulses	30..42	b110,b113,..,b146
	RPE pulses	47..59	b166,b169,..,b202
	RPE pulses	64..76	b222,b225,..,b258
	Log.area ratio 2,6	2,6	b7,b27

Table A2.1b. Subjective importance of encoded bits (the parameter and bit numbers refer to table 1.1)

ANNEX 3. FORMAT FOR TEST SEQUENCE DISTRIBUTION

A3.1. TYPE OF FILES PROVIDED

Three types of files are provided:

- Files for input of the encoder : SEQxx.INP
- Files for input of decoder or comparison with encoder output: : SEQxx.COD
- Files for comparison with the decoder output : SEQxx.OUT

Two diskettes are provided containing all the digital test sequences.

The 1st flexible disk contains the SEQ01.INP, SEQ01.COD, SEQ01.OUT, SEQ02.INP, SEQ02.COD, SEQ02.OUT files. The 2nd flexible disk contains the SEQ03.INP, SEQ03.COD, SEQ03.OUT, SEQ04.INP, SEQ04.COD, SEQ04.OUT, SEQ05.COD, SEQ05.OUT files. Table A3.1 gives the contents of the two distribution flexible disks and also the size in bytes and the number of frames for each test sequence file.

A3.2. FILE FORMAT DESCRIPTION

All the files are written in binary using 16 bit words. This means that input samples (sop[k]), output samples (srop[k]) and coded parameters use 2 bytes each. Hence the sizes of the files are directly related to the number of processed frames.

For files with .INP or .OUT extension type:

Size (in bytes) = No of frames * 160 * 2 ;

For files with .COD extension type:

Size (in bytes) = No of frames * 76 * 2;

Table A3.1 shows the size of all the files written in direct binary format.

The diskette is formatted according to the high capacity (1.2 Mb) specifications for MS/DOS PC-AT compatible computers.

DISK No	FILE	FRAMES	SIZE (bytes)
1	SEQ01.INP SEQ01.COD SEQ01.OUT	584	186880 88768 186880
1	SEQ02.INP SEQ02.COD SEQ02.OUT	947	303040 143944 303040
2	SEQ03.INP SEQ03.COD SEQ03.OUT	673	215360 102296 215360
2	SEQ04.INP SEQ04.COD SEQ04.OUT	520	166400 79040 166400
2	SEQ05.COD SEQ05.OUT	64	9728 20480

Table A3.1. <u>Contents of diskettes and size of files</u>

Speech Codec for the European Mobile Radio System

S6.1

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Philips
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C. Galand, M. Rosso
IBM
Research Laboratories
La Gaude, France

ABSTRACT

In 1991 a digital mobile radio system will be introduced in Europe. The speech codec to be used as the standard will be presented. The coding scheme which has been selected by the CEPT Groupe-Speciale-Mobile (GSM) as a result of formal subjective listening tests, is based on the Regular-Pulse Excitation LPC technique (RPE-LPC) combined with Long-Term Prediction (LTP). The so-called RPE-LTP codec has a net bit rate of 13 kbit/s. The algorithm and the experimental implementations based on different VLSI signal processors will be decribed and demonstrated by tape recordings.

1. INTRODUCTION

In the context of the standardisation of the future Pan-European digital mobile radio system, the CEPT Groupe Speciale Mobile (GSM) has carried out recently subjective codec tests using different proposals /1/. As a result of these tests, a Regular-Pulse Excitation LPC scheme was preselected /2/ on the basis that it produced the highest average speech quality. A conclusion from these tests was that the speech quality during transmission with errors could be even further improved. This led to the 'Franco-German Compromise Codec' which combines the features of the preselected German proposal /2/ with features of the French proposal /3/. The resulting RPE-LTP codec /4/ will be recommended as the final CEPT standard.

The basic RPE-scheme is related firstly to the well known baseband RELP-codec (RELP = Residual Excited Linear Prediction, /5/) and secondly to the Multi-Pulse-Excitation-LPC technique (MPE-LPC, /6/). The advantage of the baseband RELP-codec is its relative low complexity whereas the speech quality is limited due to the tonal noise which is introduced by the process of high frequency regeneration. In contrast to this, the MPE-LPC technique produces excellent speech quality but the complexity is rather high. A compromise between both techniques is the Regular-Pulse Excitation LPC (RPE-LPC, /7/), especially in its simplified version (/7/, /2/).

2. GSM CODEC TEST /1/

Initially, more than 20 different codec proposals from 9 European countries were under consideration. For the international formal listening tests this number was reduced by national tests to 6 codecs from 6 countries. The gross bit rate including error protection was 16 kbit/s. After an initial evaluation of the test results, two sub-band codecs were withdrawn. Thus, in the final test evaluation two sub-band codecs and two residual excited LPC codecs were included:

RPE-LPC: Regular-Pulse Excitation
Linear Predictive Coding
Fed. Rep. of Germany / Philips

MPE-LTP: Multi-Pulse Excitation
Long-Term Prediction
France / IBM

SBC-APCM: Sub-Band Coding
block adaptive PCM in 14 sub-bands
Sweden / ELLEMTEL

SBC-ADPCM Sub-Band Coding
adapt. different. PCM in 6 sub-bands
England / British Telecom Research

The codecs were tested in 7 languages under various transmission conditions:

* 3 input levels: 12, 22, 32 dB below overload
* 3 bit error rates: 0, 1%, 0.1% (random)
* 1 and 2 transcodings
* 2 different forms of environmental noise

Several reference conditions were included, such as a companded FM with carrier to noise ratios of 18 and 26 dB combined with simulated fading at a vehicle speed of 10 m/s (36 km/h).

The average speech quality taken over all test conditions in terms of the five-point mean opinion score (MOS), the net bit rates, and the computational complexities are given in table 1.

Codec	Speech Quality	Net Bit Rate kbit/s	Complexity MOPs/s
RPE-LPC	3.54	14.77	1.5
MPE-LTP	3.27	13.20	4.9
SBC-APCM	3.14	13.00	1.5
SBC-ADPCM	2.92	15.00	1.9
FM	1.95		

Table 1: Result of the codec test
Quality: 1 = bad 5 = excellent

CH2561-9/88/0000-0227 $1.00 © 1988 IEEE

It is noted that each of the codecs exceeded the speech quality of the analogue reference system (FM). The RPE-LPC codec obtained the highest average quality score. The analysis of the individual test conditions revealed that the quality advantage of the RPE codec decreased with an increasing bit error rate while the quality of the MPE-LTP codec was not very much affected. This motivated the 'Franco-German Codec Team' to investigate possibilities of combining the features of the RPE-LPC codec /2/ with features of the MPE-LTP codec /3/. The most important modification was the addition of a long-term prediction loop (LTP). The result was that the net bit rate could be reduced from 14.77 kbit/s to 13.0 kbit/s while maintaining the same level of quality. This increased the bit-rate capacity to improve error protection. The modified scheme is called RPE-LTP codec /4/.

3. BLOCK DIAGRAM OF THE ENCODER

The block diagram of the encoder is subdivided in 5 sections as shown in figure 1 /9/.

Preprocessing: Offset compensation is applied, to prevent a DC component being translated into an annoying side tone by the process of high frequency regeneration in the decoder. A first order FIR preemphasis filter is used for numerical reasons.

LPC Analysis: In the segmentation buffer, the speech signal is divided into non-overlapping segments having a length of T_0 = 20 ms (160 samples). A new LPC analysis is performed for each segment by calculating 8 reflection coefficients r(i) using the Schur recursion algorithm /8/. Due to the favourable quantisation characteristics, the reflection coefficients are converted into Log.Area Ratios (LARs). A piecewise linear approximation is utilised. Due to their different dynamic ranges and different asymmetric amplitude distributions, the transformed coefficients LAR(i) have different limits and are quantised uniformly as shown in table 2.

LAR No. i	1&2	3&4	5&6	7&8
bits/LAR	6	5	4	3

Table 2: Bit assignment of the LAR coefficients

Short-Term Analysis Filtering: The 8 coefficients of the short-term analysis filter are preprocessed as follows: First, the quantised and coded Log.-Area-Ratios are decoded. Then the most recent and the previous set of LAR coefficients are interpolated linearly within a transition period of 5 ms to avoid spurious transients. Finally, the interpolated Log.-Area Ratios are reconverted into the coefficients r'(i) of the FIR lattice filter.

The computation cycle outlined so far is repeated every 20 ms and produces 160 samples of the prediction error signal d.

Long-Term Predictor Loop: The LTP loop is used to compute the estimate d" of the residual signal d from the reconstructed excitation signal e'. The LTP filter is characterised by the gain b and the delay N according to

$$d''(k) = b' \cdot d'(k-N) \qquad (1)$$

where b' denotes the quantised versions of b. The parameters b and N are calculated every 5 ms (40 samples). Each segment of the 160 samples of the residual d, beginning with $d(k_0)$, is subdivided into four sub-segments $d(k_0+j\cdot 40+i)$ (j=0,1,2,3; i=0,..39). Then the cross-correlation functions R(λ) is calculated according to

$$R(\lambda) = \sum_{i=0}^{39} d(k_j+i)\cdot d'(k_j+i-\lambda); \quad k_j=k_0+j\cdot 40 \quad j = 0,1,2,3 \quad \lambda = 40..120 \qquad (2)$$

The optimum delay value N then is searched, for which this function has its maximum

$$R(N) = \max \{ R(\lambda);\ \lambda = 40...120 \}. \qquad (3)$$

Due to the lower limit of λ = 40, N does not necessarily correspond to one pitch period of the speech signal, but at least to a multiple of this period. The long-term predictor gain b for the j-th sub-segment is calculated as

$$b = R(N) \Big/ \sum_{i=0}^{39} d'^2(k_j+i-N). \qquad (4)$$

The LTP parameters b and N are encoded with 2 and 7 bits, respectively.

RPE Encoding: A FIR 'block filter' algorithm is applied to each sub-segment of 40 samples of the second residual signal e. Only the 40 centrical samples of the convolution of the 40 input samples with the 11-tap impulse response are calculated. For notational convenience the block filtered version of each sub-segment is denoted by x(k), k=0...39. For the next step the filtered signal x is down-sampled by a ratio of 3 resulting in three interleaved sequences of lengths 14, 13, and 13, which are divided again into 4 sequences xm of length 13

$$x_m(i) = x(k_j+m+3\cdot i); \quad m = 0,1,2,3 \quad i = 0,1..12 \quad k_j = k_0 + j\cdot 39 \qquad (5)$$

with k_j defining the beginning of the j-th sub-segment and with m denoting the phase of the decimation grid. The optimum candidate sequence $x_M(i)$ is selected being the one with the maximum energy /7,2/

$$E = \max_m \sum_{i=0}^{39} x_m^2(i); \qquad m=0,1,2,3. \qquad (6)$$

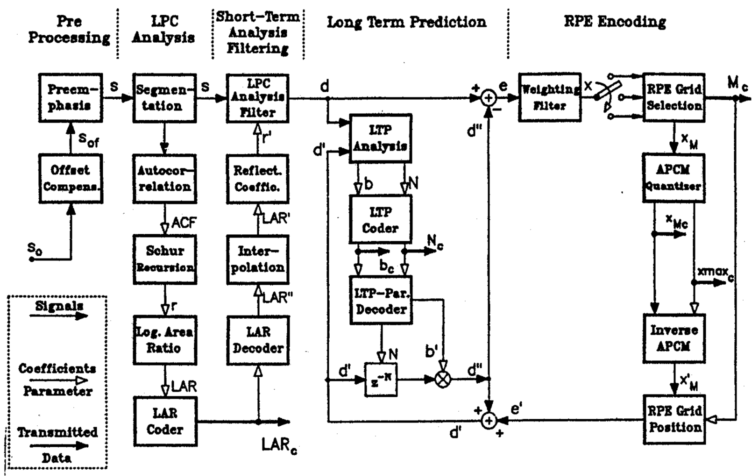

Figure 1: Block diagram of the RPE-LTP encoder (b_c, b' = coded and quantised version of b, respectively)

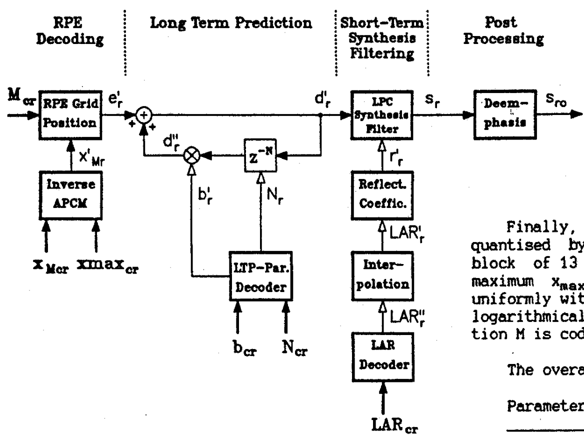

Figure 2: Block diagram of the RPE-LTP decoder
(br' = received/quantised version of b)

Thus the explicit solution of the RPE-approximation of the prediction error signal e requires only energy calculations and can be interpreted as a generalisation of the sample rate decimation process in a baseband RELP-coder.

Finally, the selected RPE-sequence $x_M(i)$ is quantised by block adaptive PCM (APCM). Each block of 13 samples is normalised by its block maximum x_{max}. The samples are then quantised uniformly with 3 bits, the block maximum is coded logarithmically with 6 bits, and the grid position M is coded with 2 bits.

The overall bit allocation is given in tab. 3

Parameter		Number of bits
8 LPC coefficients	LAR(i)	36
4 LTP coefficients	b	8
4 LTP delays	N	28
4 RPE grids	M	8
4 block maxima	x_{max}	24
52 RPE samples	x_M	156
bits per frame (20 ms)		260

Table 3: Bit allocation (bit rate = 13.0 kbit/s)

4. BLOCK DIAGRAM OF THE DECODER

The decoder is shown in fig. 2. The RPE parameters M_{cr}, x_{Mcr}, and x_{maxcr} are decoded and used to reconstruct the excitation e_r' of the long-term synthesis filter which produces the excitation signal d'_r for the short-term synthesis filter. The sample rate of the denormalised RPE samples x_{Mr}' is increased by a factor of 3 by inserting zero samples and by placing the non-zero samples in the correct temporal grid position M.

5. ERROR PROTECTION

Due to the radio transmission scheme utilising time division multiplex, GMSK and (optionally) frequency hopping, the gross bit rate including error protection is 22.8 kbit/s. At the time when this paper was written, different versions of the error protection scheme were in the discussion. Convolutional codes and Viterbi decoding will be used. The frame of 260 bits produced every 20 ms by the speech encoder will probably be sub-divided into 3 classes according to the different bit error sensitivities. These three classes will be protected differently. From preliminary listening tests based on simulations of the error protection schemes, it can be concluded that the RPE-LTP codec is fairly tolerant to errors. At a bit error rate on the radio channel of 5%, no severe degradation of the speech quality was recognised.

6. IMPLEMENTATIONS

The codec algorithm will be specified on bit level as an obligatory recommendation. Meanwhile, several implementations have been completed, using different 16 bit VLSI signal processors as shown in table 4.

processor: implementation by:	PCB5011 Philips	DSP-16 IBM	TMS320C25 CNET /10/
instruction cycle	125 ns	100 ns	100 ns
computational load	60 %	40 %	45 %
program memory	2 K	2 K	3 K
external data RAM	1 K	1 K	1 K

Table 4: Comparison of the codec implementation

For the implementation using the Philips signal processor PCB5011, a signal delay of only 28 ms has been measured.

The DSP-16 is an IBM proprietary processor which has been described in /11/.

7. CONCLUSIONS

The specification of the speech codec to be used as the standard in the Pan-European digital mobile radio system is almost complete. The net and the gross bit rates are 13 and 22.8 kbit/s, respectively. The speech quality is far superior than that obtained from present day analogue mobile radio systems. The speech coding algorithm can be implemented using just a single 16 bit VLSI signal processor with external data memory of about 1Kx16 bits.

Acknowledgements: Many colleagues have contributed to this work. The authors would like to mention especially H. Bauer, W. Koch, D. Lorenz, P. Patrick (Philips, Nuernberg), D. Aubert, Ph. Elie, J. Paturet (IBM, La Gaude), H.J. Braun (FTZ, Darmstadt), P. Combescure, J.P. Petit, and Mdm. D. Massaloux (CNET, Lannion).

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/2/ K.Hellwig, R.Hofmann R.J.Sluyter, P.Vary: MATS-D Speech Codec: Regular-Pulse Excitation LPC. Special Issue 'SPEECH COMMUNICATION' on Medium Bit Rate Coding, No. 1, 1988

/3/ C. Galand, E. Lancon, M. Rosso, Ph. Elie: MPE/LTP Algorithm for Mobile Radio Applications. 'SPEECH COMMUNICATION', No. 1, 1988

/4/ P. Vary, R.J. Sluyter, C. Galand, M. Rosso: RPE-LPC Codec - The Candidate for the GSM Radio Communication System. International Conf. on Digital Land Mobile Radio Communic. 30.06.-03.07.1987, Venice, Italy, pp 507-516

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/8/ J. Schur: Über Potenzreihen, die im Innern des Einheitskreises beschränkt sind. Journal für die reine und angewandte Mathematik, 1917, vol. 147, pp.205-232

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/10/ P. Combescure, J.P. Petit: personal communication . December 1987

/11/ C. Galand, C. Couturier, G. Platel, R. Vermot-Gauchy: Voice excited predictive coder (VEPC) implementation on a 10 MIPS signal processor. IBM Journal of Research and Development, March 1985

1811

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

00000 U.S. PTO
03/04/97

Application Serial No. 08/540,318
Filing Date .... October 11, 1995
Inventor.. Ferriere
Applicant .... Microsoft Corporation
Group Art Unit 2603
Examiner..
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission Over A Network

CHANGE OF CORRESPONDENCE AND FEE ADDRESS
PURSUANT TO 37 CFR 1.33(D)

RECEIVED
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GROUP

97 MAR 18 PM 3:30
GROUP 2

To: Commissioner of Patents and Trademarks
Washington, D.C. 20231

Dear Sir:

Please change the address of the attorneys of record in this case and direct all further communications to:

Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201
Telephone: (509)324-9256
Fax: (509)323-8979

Original notification of this address change was previously filed with the Office of Enrollment and Discipline, and was properly executed. A copy of the original notification is attached, pursuant to Section 601.03 of the MPEP.

I hereby certify the items listed above as enclosed are being deposited with the U.S. Postal Service as first class mail in an envelope addressed to Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below indicated date.

Date: 2/28/97 By: Helen M. Hare
Helen M. Hare

LEE & HAYES, PLLC
INTELLECTUAL PROPERTY LAW

W. 201 NORTH RIVER DRIVE, SUITE 430
SPOKANE, WASHINGTON 99201 U.S.A.
(509) 324-9256 [TELEPHONE]
(509) 323-8979 [FACSIMILE]

November 18, 1996

00000 U.S. PTO
03/04/97

Office of Enrollment and Discipline
U.S. Patent and Trademark Office
Box OED
Washington, D.C. 20231

Re: Change of Address
Lewis C. Lee
Registration No. 34,656

97 MAR 19 PM 3:30
GROUP 260

Dear Sir:

Pursuant to 37 CFR 10.11, this is to inform you of my new address:

Lewis C. Lee
Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201

Please make this entry on the Register of Attorneys and Agents and confirm receipt of this change of address.

Sincerely,

Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

UNITED STATES DEP[illegible]MENT OF COMMERCE
Patent and Trademar[illegible]fice
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO.
08/540,818	10/11/95	FERRIERE	P MS1069US

EXAMINER

26M1/0922

LEE & HAYES
W 1818 FRANCIS #160
SPOKANE WA 99205

JUNG, M

ART UNIT	PAPER NUMBER
2603	7

DATE MAILED: 09/22/97

This is a communication from the examiner in charge of your application.
COMMISSIONER OF PATENTS AND TRADEMARKS

OFFICE ACTION SUMMARY

☒ Responsive to communication(s) filed on October 11, 1995.

☐ This action is **FINAL**.

☐ Since this application is in condition for allowance except for formal matters, **prosecution as to the merits is closed** in accordance with the practice under *Ex parte Quayle*, 1935 D.C. 11; 453 O.G. 213.

A shortened statutory period for response to this action is set to expire three month(s), or thirty days, whichever is longer, from the mailing date of this communication. Failure to respond within the period for response will cause the application to become abandoned. (35 U.S.C. § 133). Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

Disposition of Claims

☒ Claim(s) 1-36 is/are pending in the application.
Of the above, claim(s) 29-36 is/are withdrawn from consideration.
☐ Claim(s) ________ is/are allowed.
☒ Claim(s) 1-28 is/are rejected.
☐ Claim(s) ________ is/are objected to.
☒ Claim(s) 1-36 are subject to restriction or election requirement.

Application Papers

☐ See the attached Notice of Draftsperson's Patent Drawing Review, PTO-948.
☐ The drawing(s) filed on ________ is/are objected to by the Examiner.
☐ The proposed drawing correction, filed on ________ is ☐ approved ☐ disapproved.
☐ The specification is objected to by the Examiner.
☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. § 119

☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d).

☐ All ☐ Some* ☐ None of the CERTIFIED copies of the priority documents have been

☐ received.
☐ received in Application No. (Series Code/Serial Number) ________.
☐ received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

*Certified copies not received: ________.

☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e).

Attachment(s)

☒ Notice of Reference Cited, PTO-892
☒ Information Disclosure Statement(s), PTO-1449, Paper No(s). 6
☒ Interview Summary, PTO-413
☐ Notice of Draftperson's Patent Drawing Review, PTO-948
☐ Notice of Informal Patent Application, PTO-152

--SEE OFFICE ACTION ON THE FOLLOWING PAGES--

PTOL-326 (Rev. 9/96)

★ U.S. GPO: 1996-404-496/40517

1. Restriction to one of the following inventions is required under 35 U.S.C. 121:

   I. Claims 1-28, drawn to communication system and method for adaptively encoding data depending on the effective bit rate of receiving modem , classified in class 370, subclass 465.

   II. Claims 29-36, drawn to encoder including quantizers, classified in class 375, subclass 245.

2. The inventions are distinct, each from the other because of the following reasons: Inventions I and II are related as subcombinations disclosed as usable together in a single combination. The subcombinations are distinct from each other if they are shown to be separately usable. In the instant case, invention I has separate utility such as an adaptive encoding scheme without having to use the specific encoder as claimed in invention II. See MPEP § 806.05(d).

3. Because these inventions are distinct for the reasons given above and have acquired a separate status in the art as shown by their different classification, restriction for examination purposes as indicated is proper.

4. Because these inventions are distinct for the reasons given above and the search required for Group I is not required for Group II, restriction for examination purposes as indicated is proper.

5. Because these inventions are distinct for the reasons given above and have acquired a separate status in the art because of their recognized divergent subject matter, restriction for examination purposes as indicated is proper.

6. During a telephone conversation with Mr. Lewis Lee on September 9, 1997 a provisional election was made with traverse to prosecute the invention of I, claims 1-28. Affirmation of this election must be made by applicant in responding to this Office action. Claims 29-36 are withdrawn from further consideration by the examiner, 37 CFR 1.142(b), as being drawn to a non-elected invention.

Claim Rejections - 35 USC § 112

7. Claims 1-28 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Throughout the claims, it is not clear whether or not the "block size" means the number of bits in a unit data block which is to be encoded.

In claim 2, it is not clear whether other block sizes are selected to encode some of the data at different rate, or other block sizes are selected to parallel encode at a different rate the same data already encoded at one rate.

In claim 3, again, the indefiniteness arises from the unclear meaning of "block size". Is the claim recitation trying to define that different number of bits per block constitutes different block sizes?

In claim 5, it is not clear how the "altering" step and the "modifying" step are tied to the rest of the claim limitations; in other words, how are the steps related to the modem rates? Are those steps performed when the communication with a modem operating at a different rate is being initiated?

In claim 6, it is not clear what is the significance of transmitting the data block at the encoded bit stream bit rate (isn't it the most conventional thing to do?).

The indefiniteness mentioned above apply to some of the remaining claims which will not be repeated.

Claim Rejections - 35 USC § 103

8. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

9. Claims 1-4, 6-7, 10-18, 22, 24-28 are rejected under 35 U.S.C. 103(a) as being unpatentable over Li, 5,309,562.

Li discloses a method and apparatus for establishing protocol spoofing from a modem. Li teaches that the initiating modem appends a modem spoofing initiation protocol word containing modem limitations to the initial negotiation frame and sends the frame to the receiving modem. The receiving modem compares its own limits with the received limits and takes the lower limit as the agreement. Both the initiating modem and the receiving modem use the agreed limits to examine the maximum block size and window size. If the block size and window size fall within the agreed limits, the spoofing service begins. See column 3, lines 33-52.

Li teaches selecting a block size for data block since the negotiation process is performed for determining the common communication parameters including the block size, between two communication sites. Li fails a specific teaching of applying the invention in audio communication environment. However, in digital communication technique, digitized audio is treated much the same way as other digital non-audio data. Therefore, it would have been obvious for one of ordinary skill in the art at the time of the invention to apply the teaching of Li in audio data servicing environment for determining encoding parameters depending on the capabilities of the receiving modem. Li also fails to teach various applications such as transmitting over a distribution network, over a communication network, and the network being a wireless communication network, or a wire-based data network. It would have been obvious for one of ordinary skill in the art to apply the disclosed teaching to various known communications environment such as the ones specified above since the principal does not have to be altered when applying it to different applications.

Allowable Subject Matter

10. Claims 5, 8-9, 19-21, 23 would be allowable if rewritten to overcome the rejection(s) under 35 U.S.C. 112 set forth in this Office action and to include all of the limitations of the base claim and any intervening claims.

Conclusion

11. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Li et al., 5,617,423, discloses a communication technique of sending voice over data with a feature of negotiating data communication parameters including speech compression parameter.

Jacobsmeyer, 5,541,955, discloses an adaptive data rate modem including adaptive data rate encoder supporting multiple data rates.

12. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Min Jung whose telephone number is (703) 305-4363. The examiner can normally be reached on M-F from 9AM to 5PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Douglas Olms, can be reached on (703) 305-4703. The fax phone number for this Group is (703) 308-9051.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3900.

M.JUNG

September 18, 1997

Min Jung

Min Jung

TO SEPARATE, HOLD TOP AND BOTTOM EDGES, SNAP–APART AND DISCARD CARBON

FORM PTO-892 (REV. 2-92)

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE

NOTICE OF REFERENCES CITED

SERIAL NO.	GROUP ART UNIT	ATTACHMENT TO PAPER NUMBER
08/540,818	2603	7

APPLICANT(S): Ferriere

U.S. PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	NAME	CLASS	SUB-CLASS	FILING DATE IF APPROPRIATE
	A	5309562	5-1994	Li	395	200.67	
	B	5617423	4-1997	Li et al.	370	426	12-1993
	C	5541955	7-1996	Jacobsmeyer	375	222	11-1992
	D						
	E						
	F						
	G						
	H						
	I						
	J						
	K						

FOREIGN PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	COUNTRY	NAME	CLASS	SUB-CLASS	PERTINENT SHTS. DWG	PERTINENT PP. SPEC.
	L								
	M								
	N								
	O								
	P								
	Q								

OTHER REFERENCES (Including Author, Title, Date, Pertinent Pages, Etc.)

R	
S	
T	
U	

EXAMINER	DATE
Min Jung	9/18/97

* A copy of this reference is not being furnished with this office action.
(See Manual of Patent Examining Procedure, section 707.05 (a).)

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKETT NO.

EXAMINER	

ART UNIT	PAPER NUMBER
	7

DATE MAILED:

EXAMINER INTERVIEW SUMMARY RECORD

All participants (applicant, applicant's representative, PTO personnel):

(1) Lewis Lee (3)

(2) Min Jung (4)

Date of interview September 9, 1997

Type: ☒ Telephonic ☐ Personal (copy is given to ☐ applicant ☐ applicant's representative).

Exhibit shown or demonstration conducted: ☐ Yes ☐ No. If yes, brief description:

Agreement ☐ was reached with respect to some or all of the claims in question. ☐ was not reached.

Claims discussed: 1-36

Identification of prior art discussed:

Description of the general nature of what was agreed to if an agreement was reached, or any other comments: In response to the examiner's Restriction requirement, Mr. Lewis elected Group I, claims 1-28, directed to communication system which adapts its performance to the modem rate. The election was made with traverse.

(A fuller description, if necessary, and a copy of the amendments, if available, which the examiner agreed would render the claims allowable must be attached. Also, where no copy of the amendments which would render the claims allowable is available, a summary thereof must be attached.)

☐ 1. It is not necessary for applicant to provide a separate record of the substance of the interview.

Unless the paragraph below has been checked to indicate to the contrary, A FORMAL WRITTEN RESPONSE TO THE LAST OFFICE ACTION IS NOT WAIVED AND MUST INCLUDE THE SUBSTANCE OF THE INTERVIEW (e.g., items 1-7 on the reverse side of this form). If a response to the last Office action has already been filed, then applicant is given one month from this interview date to provide a statement of the substance of the interview.

☐ 2. Since the examiner's interview summary above (including any attachments) reflects a complete response to each of the objections, rejections and requirements that may be present in the last Office action, and since the claims are now allowable, this completed form is considered to fulfill the response requirements of the last Office action. Applicant is not relieved from providing a separate record of the substance of the interview unless box 1 above is also checked.

Min Jung
Examiner's Signature

PTOL-413 (REV. 2 -93)

ORIGINAL FOR INSERTION IN RIGHT HAND FLAP OF FILE WRAPPER

12-17-97
C. Aymace

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No. 08/540,818
Filing Date ....October 11, 1995
Inventor..Ferriere
Applicant .... Microsoft Corporation
Group Art Unit 2731
Examiner ....M. Jung
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission Over a Network

8
1-22-98
D. Little

RESPONSE TO OFFICE ACTION DATED SEPTEMBER 22, 1997

To: Commissioner of Patents and Trademarks
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201

1/4/98
Cr

In the Claims:

Claims 1-36 were presented in the original application.

Claims 29-36 have been withdrawn from consideration.

Kindly cancel claims 1-4, 7, and 15-18 without prejudice.

Claims 5-6, 8-13, and 19-23 are amended.

Claims 14 and 24-28 remain unchanged.

New claims 37-48 are added.

Claims 5-6, 8-14, 19-28, and 37-48 are pending.

A

Please amend claims 5-6, 8-13, and 19-23 as follows:

1 ~~5~~. (Amended) A method for encoding digital audio data to be transmitted to a modem operating at an effective bit rate, the method comprising the following steps: [as recited in claim 4 further comprising the following additional steps:]

sampling audio data at a first input sampling rate to produce audio samples;

encoding the audio samples into audio data blocks of a first block size selected from among a set of available block sizes, the block size representing a number of data bits contained within an individual audio data block, wherein the first block size and the first input sampling rate provide a first encoded bit stream bit rate that is less than or equal to the effective bit rate of the modem;

subsequently altering at least one of the first block size to a second block size or the first input sampling rate to a second input sampling rate; and

[modifying] providing [the] a second encoded bit stream bit rate as a result of the altering step by performing at least one of the following two steps: (1) sampling the audio data at the [altered] second input sampling rate or (2) encoding the audio samples into audio data blocks having the second [of a different] block size.

2 ~~6~~. (Amended) A method as recited in claim [4] 1 ~~5~~ further comprising the step of initially transmitting the audio data blocks at the first encoded bit stream bit rate and subsequently transmitting the audio data blocks at the second encoded bit stream bit rate.

3 ~~8.~~ (Amended) A method for supplying digital audio files to a recipient, the recipient having a modem operating to receive digital data at an effective bit rate, the audio files being configured into individual audio data blocks wherein each audio data block contains a certain number of bits of digital audio data sampled at an input sampling rate, the method comprising the following steps: [as recited in claim 7 further comprising the following additional steps:]

storing multiple versions of [the] an audio file, each version of the audio file being [that are] configured in audio data blocks of different block sizes and produced using different input sampling rates, wherein the block size represents a number of data bits contained within an individual audio data block; [and]

choosing an appropriate version of the audio file that has a block size and an input sampling rate which produces [the] a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem; and

transmitting the audio data blocks at the bit stream bit rate to the recipient's modem.

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7 ~~9.~~ (Amended) A method for supplying digital audio files to a recipient, the recipient having a modem operating to receive digital data at an effective bit rate, the method comprising the following steps: [as recited in claim 7 further comprising the following additional steps:]

storing [the] an audio file in uncompressed format; [and]

configuring [an] the audio file into individual audio data blocks in real-time wherein each audio data block contains a selected number of bits of digital audio data sampled at a selected input sampling rate, the selected number of bits and the selected input sampling rate determining a bit stream bit rate that is less than or

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equal to the effective bit rate of the recipient's modem; and [prior to the transmitting step]

transmitting the audio data blocks at the bit stream bit rate to the recipient's modem.

4 ~~10~~. (Amended) A method as recited in claim [7] ~~8~~ 3 wherein the transmitting step comprises transmitting the audio data blocks over a distribution network.

5 ~~11~~. (Amended) A method as recited in claim [7] ~~8~~ 3 wherein the transmitting step comprises transmitting the audio data blocks over a communication network.

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6 ~~12~~. (Amended) A method as recited in claim [7] ~~8~~ 3 further comprising the steps of receiving the audio data blocks at the recipient and reproducing audio sound from the audio data blocks as they are received.

11 ~~13~~. (Amended) A method for transmitting multiple digital audio files concurrently to a recipient, the recipient having a modem operating to receive digital data at an effective bit rate, the method comprising the following steps:

performing for each digital audio file the following steps:

configuring the audio file into individual audio data blocks, each audio data block containing a certain number bits of digital audio data sampled at an input sampling rate;

selecting a block size for the audio data blocks from among a set of available block sizes and an input sampling rate from among a set of available input sampling rates that determine a bit stream bit

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rate, the block size representing the number of bits of digital audio data in an individual audio data block;

said selecting step selecting the block size and the input sampling rate for each audio file which ensure that a total bit stream bit rate made up of combined bit stream bit rates of all digital audio files is less than or equal to the effective bit rate of the recipient's modem; and

transmitting the audio data blocks for each audio file at the bit stream bit rate to the recipient's modem.

---

13 ~~19~~. (Amended) An audio file distribution system [as recited in claim 16 wherein:] comprising:

a client computing unit having a modem operating to receive digital data at an effective bit rate;

[the] an audio file database [stores] to store multiple versions of [each] an audio file, each version of the audio file being [that are] configured in audio data blocks of different block sizes and produced using different input sampling rates, wherein block size represents a number of data bits contained within an individual audio data block; and

[the] an audio server [selects] to select an appropriate version of the audio file that has a block size and an input sampling rate which produces [the] a bit stream bit rate that is less than or equal to the effective bit rate of the client's modem, the audio server retrieving the appropriate version from the audio file database and supplying the audio file to the client computing unit.

15 ~~20~~. (Amended) An audio file distribution system [s recited in claim 16 wherein:] comprising:

a client computing unit having a modem operating to receive digital data at an effective bit rate;

[the] an audio file database [stores the] to store an audio [files] file in uncompressed format; and

[the] an audio server [is configured] to encode the audio file into individual audio data blocks in real-time wherein each audio data block contains a selected number of bits of digital audio data sampled at a selected input sampling rate, the selected number of bits and the selected input sampling rate determining a bit stream bit rate that is less than or equal to the effective bit rate of the client's modem.

17 ~~21~~. (Amended) An audio file distribution system [as recited in claim 16 wherein] comprising:

a client computing unit having a modem operating to receive digital data at an effective bit rate;

an audio file database to store an audio file; and

[the] an audio server to supply the audio file as individual audio data blocks which contain a certain number of bits of digital audio data that have been sampled at a selected sampling rate, the audio server [has] having a size/rate lookup table listing various combinations of [block sizes] the number of bits and the sampling rates indexed with associated encoded bit stream bit rates, [and] the audio server [is] being configured to select the [block size] number of bits and the sampling rate

using the size/rate lookup table so that the associated encoded bit stream bit rate is less than or equal to the effective bit rate of the client's modem.

14 ~~22~~. (Amended) An audio file distribution system as recited in claim [16] ~~19~~ 13 wherein the client computing unit decodes the audio data blocks and reproduces audio sound therefrom as the audio data blocks are received from the audio server.

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19 ~~23~~. (Amended) An audio file distribution system ~~as recited in claim 16~~ comprising ~~wherein:~~

a client computing unit having a modem operating to receive digital data at an effective bit rate;

an audio file database to store a plurality of audio files; and

an audio server to retrieve multiple audio files from the audio file database and encode the audio files as individual audio data blocks which contain a certain number bits of digital audio data that have been sampled at a selected sampling rate wherein the number of bits of digital data and the sampling rate are selected to provide an encoded bit stream bit rate that is less than or equal to the effective bit rate of the client's modem; and

[the audio server encodes multiple digital audio files into audio data blocks; and]

the client computing unit is configured to decode the audio data blocks into audio frames, mix the audio frames, and reproduce sound from the audio frames.

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Please add new claims 37-48 as follows:

8 --~~37~~. A method as recited in claim 7 ~~9~~ wherein the transmitting step comprises transmitting the audio data blocks over a distribution network.--

9 --~~38~~. A method as recited in claim 7 ~~9~~ wherein the transmitting step comprises transmitting the audio data blocks over a communication network.--

10 --~~39~~. A method as recited in claim 7 ~~9~~ further comprising the steps of receiving the audio data blocks at the recipient and reproducing audio sound from the audio data blocks as they are received.--

16 --~~40~~. An audio file distribution system as recited in claim 15 ~~20~~ wherein the client computing unit decodes the audio data blocks and reproduces audio sound therefrom as the audio data blocks are received from the audio server.--

18 --~~41~~. An audio file distribution system as recited in claim 17 ~~21~~ wherein the client computing unit decodes the audio data blocks and reproduces audio sound therefrom as the audio data blocks are received from the audio server.--

25 --~~42~~. An audio file serving system for serving an audio file to a recipient having a modem that receives digital data at an effective bit rate, comprising:

an audio file database to store multiple versions of an audio file, each version of the audio file being configured in audio data blocks having different block sizes and produced using different input sampling rates, wherein block size

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represents a number of data bits contained within an individual audio data block; and

an audio server to select an appropriate version of the audio file that has a block size and an input sampling rate that produces a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem, the audio server retrieving the appropriate version from the audio file database and supplying the audio file to the recipient.--

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--~~43~~. An audio file serving system for serving an audio file to a recipient having a modem that receives digital data at an effective bit rate, comprising:

an audio file database to store an audio file in uncompressed format; and

an audio server to encode the audio file into individual audio data blocks in real-time wherein each audio data block contains a selected number of bits of digital audio data sampled at a selected input sampling rate, the selected number of bits and the selected input sampling rate determining a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem.--

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--~~44~~. An audio file serving system for serving an audio file to a recipient having a modem that receives digital data at an effective bit rate, comprising:

an audio file database to store an audio file; and

an audio server to supply the audio file as individual audio data blocks which contain a certain number of bits of digital audio data that have been sampled at a selected sampling rate, the audio server having a size/rate lookup table listing various combinations of the number of bits and the sampling rates that yield associated encoded bit stream bit rates, the audio server being configured to select the number of bits and the sampling rate using the size/rate lookup table so that the associated encoded bit stream bit rate is less than or equal to the effective bit rate of the recipient's modem.--

--~~45~~ 28. A method for supplying digital audio files to a recipient, the recipient having a modem to receive digital data at an effective bit rate, the audio files being configured into individual audio data blocks wherein each audio data block has a block size, which represents a number bits of digital audio data, and an input sampling rate, which represents the sample rate at which the digital audio data is sampled, the method comprising the following steps:

storing various combinations of block sizes and sampling rates in a table;

selecting a block size and a sampling rate from the table that yield an encoded bit stream bit rate is less than or equal to the effective bit rate of the recipient's modem; and

supplying the audio file at the encoded bit stream bit rate.--

--46. In a system for supplying digital audio files to a recipient having a modem to receive digital data at an effective bit rate, the audio files being stored in multiple different versions having audio data blocks of different numbers of data bits taken at different input sampling rates, a program embodied on a compute-r readable medium, comprising:

computer-executable instructions to select a version of the audio file that has an appropriate number of data bits and an input sampling rate to produce a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem; and computer-executable instructions to supply the selected version of the audio file to the recipient.--

--47. In a system for supplying digital audio files to a recipient having a modem to receive digital data at an effective bit rate, the audio files being stored in an uncompressed format, a program embodied on a compute-readable medium comprising computer-executable instructions to encode an audio file into individual audio data blocks in real-time wherein each audio data block contains a selected number of bits of digital audio data sampled at a selected input sampling rate, the selected number of bits and the selected input sampling rate determining a bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem.--

QH

--48. In a system for supplying digital audio files to a recipient having a modem to receive digital data at an effective bit rate, the audio files being configured into individual audio data blocks wherein each audio data block has a block size, which represents a number bits of digital audio data, and an input sampling rate, which represents the sample rate at which the digital audio data is sampled, a program embodied on a compute-readable medium, comprising:

computer-executable instructions to index a size/rate lookup table listing various combinations of block sizes and sampling rates and select a combination that yields an encoded bit stream bit rate that is less than or equal to the effective bit rate of the recipient's modem; and

computer-executable instructions to supply the audio file at the encoded bit stream bit rate.--

REMARKS

Applicant respectfully requests reconsideration and allowance of the subject application. Claims 5-6, 8-13, and 19-23 are amended and claims 1-4, 7, and 15-18 are canceled without prejudice. New claims 37-48 are added. Claims 5-6, 8-14, 19-28, and 37-48 are pending in this application.

Allowable Subject Matter

Claims 5, 8-9, 19-21, and 23 are indicated as allowable if rewritten in independent form and to overcome the §112 rejection (discussed below). Applicant thanks the Examiner for recognizing the patentability of these claims.

Claims 5, 8-9, 19-21, and 23 are rewritten in independent form. These claims are in condition for allowance.

Restriction Requirement

The claims have been subjected to a restriction requirement under 35 U.S.C. § 121. In particular, the Examiner has imposed the following restriction:

Invention I: Claims 1-28, drawn to a communication system and method for adaptively encoding data depending on the effective bit rate of receiving modem, classified in class 370, subclass 465.

Invention II: Claims 29-36, drawn to an encoder including quantizers, classified in class 375, subclass 245.

During a telephone conference of September 9, 1997, Applicant elected, with traverse, to prosecute Invention I, claims 1-28. This election is hereby affirmed.

Applicant respectfully traverses the restriction requirement. It is believed that the Office will search both classes when examining the elected Invention I. To support this belief, the references cited by the Examiner for Invention I are primarily classified in the two classes 370 and 375 for both Inventions I and II. Notice that Jacobsmeyer (U.S. Patent No. 5,541,955) is primarily classified in class 375, which contains Invention II. The Li et al. reference (U.S. Patent No. 5,617,423) is primarily classified in class 370, which contains Invention I. Moreover, the Li et al. reference shows the field of search including both classes 370 and 375.

As further evidence supporting withdrawal of the restriction requirement, Applicant notes that the two groups of claims contain overlapping subject matter. For instance, claim 24 defines a communication system having first and second communication units, each of which is equipped with an audio coder/decoder having multiple quantizers that encode digital audio samples. Dependent claim 27 further requires that the audio coder/decoder be a *predictive* audio coder/decoder. Claims 24 and 27 are indicated as being contained in Invention I.

On the other hand, claim 29, which is indicates as being part of invention II, defines a *predictive* audio coder having, among other things, an encoder with multiple quantizers. It would seem that for the Office to adequately search the aspects of claim 24 and 27, the Office would most likely effectively search for most of the same aspects in claim 29. Restricting the claims into two cases would therefore result in duplicative searches.

Since the same search is required for both inventions, it would prove more efficient and cost effective to examine all of the presented claims. If the restriction requirement is maintained, Applicant will experience additional fees associated with filing, prosecution, issuance, and maintenance costs. The Office will experience an additional burden of repetitive searching that can be conveniently handled in a single application.

Accordingly, Applicant respectfully requests that the restriction requirement be withdrawn.

35 U.S.C. § 112

Claims 1-28 stand rejected under 35 U.S.C. §112, second paragraph, as being indefinite. The remaining claims are amended to clarify the issues raised by the Examiner. In view of these amendments, Applicant respectfully requests that the §112 rejection be withdrawn.

35 U.S.C. § 103

Claims 1-4, 6-7, 10-18, 22, and 24-28 stand rejected under 35 U.S.C. § 103 as being unpatentable over U.S. Patent No. 5,309,562 to Li (hereinafter "Li"). Applicant respectfully traverses the rejection.

Claims 1-4 are canceled without prejudice.

Claim 6 is amended to depend from allowable claim 5, and hence is in condition for allowance.

Claim 7 is canceled without prejudice.

Claims 10-12 are amended to depend from allowable claim 8, and hence are in condition for allowance.

Claim 13 defines a method for transmitting *multiple* digital audio files *concurrently* to a recipient. The method includes "selecting the block size and the input sampling rate for each audio file which ensure that a *total* bit stream bit rate *made up of combined bit stream bit rates of all digital audio files* is less than or equal to the effective bit rate of the recipient's modem."

Li does not teach or suggest the ability to transmit multiple digital audio files concurrently, as claim 13 requires. For this reason, Applicant respectfully requests that the §103 rejection of claim 13 be withdrawn.

Claim 14 depends from claim 13 and is allowable over Li by virtue of this dependency. In addition, claim 14 requires "reproducing and mixing audio sound from the audio data blocks as they are received." This step is not taught or suggested by Li.

Claims 15-18 are canceled without prejudice.

Claim 22 is amended to depend from allowable claim 19. Thus, this claim is in condition for allowance.

Claim 24 defines a communication system having first and second communication units that are "equipped with an audio coder/decoder having multiple quantizers that encode the digital audio samples into various sized audio data blocks which contain various quantities of bits of the audio samples". Further, the first and second communication units are "using an appropriate input sampling rate and selecting an appropriate quantizer to encode the audio samples into audio data blocks that yield an encoded bit stream bit rate less than or equal to the smallest effective bit rate of the modems."

Li does not teach or suggest a communication unit that employs a codec with multiple quantizers, and selecting the appropriate quantizer to encode the samples in order to produce a bit rate that is appropriate for the receiving unit. For this reason, claim 24 is patentable over Li. Applicant respectfully requests that the §103 rejection of claim 24 be withdrawn.

Claims 25-28 depend from claim 24 and are allowable by virtue of this dependency.

__New Claims__

New claims 37-39 are identical to claims 10-12, but depend from allowable claim 9 as opposed to allowable claim 8. Accordingly, these claims are in condition for allowance.

New claims 40 and 41 are identical to claim 22, but depend from allowable claims 20 and 21, respectively.

New claims 42-44 are directed to a server system. These claims are similar to allowable claims 19-21, but are directed only to the server while omitting recitation of the client.

New claim 45 is a method claim that is similar to allowable claim 21.

New claims 46-48 are directed to a program embodied on a computer-readable medium. These claims contain limitations similar to those in allowable claims 19-21.

Conclusion

Claims 5-6, 8-14, 19-28, and 37-48 are in condition for allowance. Applicant respectfully requests reconsideration of these claim, as well as consideration of claims 29-36, and prompt issuance of the subject application.

Respectfully Submitted,

Date: Dec. 17, 1997

By: Lewis C Lee

Lewis C. Lee
Reg. No. 34,656
(509) 324-9256

GP 2731

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US

Title: System and Method for Scaleable Audio Transmission Over a Network

TRANSMITTAL LETTER AND CERTIFICATE OF MAILING

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-323-8979)
Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201

The following enumerated items accompany this transmittal letter and are being submitted for the matter identified in the above caption.

1. Transmittal letter including Certificate of Mailing
2. Response to Office Action Dated September 22, 1997
3. Return Post Card

Large Entity Status [x]   Small Entity Status [ ]

The Commissioner is hereby authorized to charge payment of fees or credit overpayments to Deposit Account No. 12-0769 in connection with any patent application processing fees under 37 CFR 1.17; and any additional filing fees under 37 CFR 1.16 for the presentation of extra claims.

Date: Dec. 17, 1997   By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

CERTIFICATE OF MAILING

I hereby certify that the items listed above as enclosed are being deposited with the U.S. Postal Service as either first class mail, or Express Mail if the blank for Express Mail No. is completed below, in an envelope addressed to The Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below-indicated date. Any Express Mail No. has also been marked on the listed items.

Express Mail No. (if applicable) ____________________

Date: Dec. 17, 1997   By: Helen M. Hare
Helen M. Hare

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.
08/540,818	10/11/95	FERRIERE P	MS1069US

LM61/0414

LEE & HAYES
W 1818 FRANCIS #160
SPOKANE WA 99205

EXAMINER	
JUNG,M	
ART UNIT	**PAPER NUMBER**
2731	9

DATE MAILED: 04/14/98

Please find below and/or attached an Office communication concerning this application or proceeding.

Commissioner of Patents and Trademarks

PTO-90C (Rev. 2/95)

☆U.S.GOVERNMENT PRINTING OFFICE 1996-411-516/40276

1- File Copy

Office Action Summary	Application No. 08/540,818	Applicant(s) Ferriere
	Examiner Min Jung	Group Art Unit 2731

—The MAILING DATE of this communication appears on the cover sheet beneath the correspondence address—

Period for Response

A SHORTENED STATUTORY PERIOD FOR RESPONSE IS SET TO EXPIRE three MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a response be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for response specified above is less than thirty (30) days, a response within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for response is specified above, such period shall, by default, expire SIX (6) MONTHS from the mailing date of this communication .
- Failure to respond within the set or extended period for response will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).

Status

☒ Responsive to communication(s) filed on December 23, 1997.

☒ This action is **FINAL.**

☐ Since this application is in condition for allowance except for formal matters, **prosecution as to the merits is closed** in accordance with the practice under *Ex parte Quayle,* 1935 C.D. 1 1; 453 O.G. 213.

Disposition of Claims

☒ Claim(s) 5-6, 8-14, 19-28, 29-36, 37-48 is/are pending in the application.

Of the above claim(s) 29-36 and 46-48 is/are withdrawn from consideration.

☒ Claim(s) 5-6, 8-14, 19-28, 37-45 is/are allowed.

☐ Claim(s) ______ is/are rejected.

☐ Claim(s) ______ is/are objected to.

~~☒~~ Claim(s) ~~29-36 and 46-48~~ are subject to restriction or election requirement.

Application Papers

☐ See the attached Notice of Draftsperson's Patent Drawing Review, PTO-948.

☐ The proposed drawing correction, filed on ______ is ☐ approved ☐ disapproved.

☐ The drawing(s) filed on ______ is/are objected to by the Examiner.

☐ The specification is objected to by the Examiner.

☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. § 119 (a)-(d)

☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 11 9(a)-(d).

☐ All ☐ Some* ☐ None of the CERTIFIED copies of the priority documents have been

☐ received.

☐ received in Application No. (Series Code/Serial Number) ______.

☐ received in this national stage application from the International Bureau (PCT Rule 1 7.2(a)).

*Certified copies not received: ______.

Attachment(s)

☐ Information Disclosure Statement(s), PTO-1449, Paper No(s). ______

☐ Notice of References Cited, PTO-892

☐ Notice of Draftsperson's Patent Drawing Review, PTO-948

☐ Interview Summary, PTO-413

☐ Notice of Informal Patent Application, PTO-152

☐ Other ______

Office Action Summary

DETAILED ACTION

Election/Restriction

1. Applicant's election with traverse of the Restriction made in Paper No. 7 is acknowledged. The traversal is on the ground(s) that both Inventions I and II require a search in class 370 and class 375. This is not found persuasive because the examiner not only based her requirement of restriction on the grounds that the search areas are different, but also on the grounds that the inventions are related as subcombinations disclosed as usable together in a single combination, and that the two inventions have acquired a separate status in the art as shown by their different classification, and also by their recognized divergent subject matter. Examiner agree with the applicants, however, that for the search to be complete, both the areas in class 370 and class 375 have to be covered. In other words, although it would not be required to search class 375 for the invention I, it would be reasonable to include the search in class 375. However, the evidence of duplicate search area alone does not justify that the inventions I and II are directed to the same invention. Claims 29-36, which are directed to an encoder having the details of encoder elements are most properly classified in class 375, subclass 245, and they are <u>different</u> from claims 1-28 which are most properly classified in class 370, subclass 465, and which are a lot more that just an encoder; e.g., communication system including an encoder.

The requirement is still deemed proper and is therefore made FINAL.

2. This application contains claims 29-36 drawn to an invention nonelected with traverse in Paper No. 8. A complete reply to the final rejection must include cancellation of nonelected claims or other appropriate action (37 CFR 1.144) See MPEP § 821.01.

3. Newly submitted claims 46-48 are directed to an invention that is independent or distinct from the invention originally claimed for the following reasons: This invention is directed to a computer program embodied on a computer-readable medium for effecting remote data accessing, classified in class 395, subclass 200.47.

Since applicant has received an action on the merits for the originally presented invention, this invention has been constructively elected by original presentation for prosecution on the merits. Accordingly, claims 46-48 are withdrawn from consideration as being directed to a non-elected invention. See 37 CFR 1.142(b) and MPEP § 821.03.

Allowable Subject Matter

4. Claims 5-6, 8-14, 19-28, 37-45 are allowed.

5. The following is a statement of reasons for the indication of allowable subject matter: Prior art fail to teach adaptation of bit rates and block sizes of audio data to be transmitted, and storage of different versions of same data corresponding to the different rates and block sizes, depending on the effective bit rates of modems.

Response to Arguments

6. Applicant's arguments filed December 23, 1997 have been fully considered but they are not persuasive. The traversal of the restriction requirement was considered. Examiner maintains her position on the restriction for the reasons provided above.

Conclusion

7. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

8. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Min Jung whose telephone number is (703) 305-4363. The examiner can normally be reached on M-F from 9AM to 5PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chi Pham, can be reached on (703) 305-4378. The fax phone number for the organization where this application or proceeding is assigned is (703) 308-9051.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is (703) 305-3900.

M.JUNG

April 9, 1998

Min Jung
MIN JUNG
PATENT EXAMINER
GROUP ~~2600~~ 2731

OIPE
MAY 15 1998

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

10/B
6/1/98

Application Serial No. 08/540,818
Filing Date ....October 11, 1995
Inventor.................... Ferriere
Applicant .... Microsoft Corporation
Group Art Unit 2731
Examiner ....M. Jung
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission Over a Network

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98 MAY 18 PM 2:08
GROUP 2700
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RESPONSE TO FINAL OFFICE ACTION DATED APRIL 14, 1998

To: Commissioner of Patents and Trademarks
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-928-2642)
Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201

In the Claims:

Claims 5-6, 8-14, and 19-48 were pending at the time of the Final Action.

Kindly cancel claims 29-36 and 46-48 without prejudice.

Claims 5-6, 8-14, 19-28, and 37-45 remain unchanged and pending in this case.

REMARKS

Applicant respectfully requests reconsideration and allowance of the subject application. Claims 29-36 and 46-48 are canceled without prejudice. Claims 5-6, 8-14, 19-28, and 37-45 are pending.

Restriction Requirement

The Examiner has made the restriction final and requires cancellation of the non-elected claims 29-36 and 46-48. In compliance with this requirement, the non-elected claims are canceled without prejudice. Applicant reserves the right to pursue these claims in a separate application.

Allowed Claims

Claims 5-6, 8-14, 19-28, and 37-45 are allowed. These claims remain unchanged and in condition for allowance.

Respectfully Submitted,

Date: May 11, 1998

By: Lewis C. Lee

Lewis C. Lee
Reg. No. 34,656
(509) 324-9256

PE
MAY 1 5 1998
JC23
PATENT & TRADEMARK OFFICE

GP
2731

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application Serial No 08/540,818
Filing Date .... October 11, 1995
Inventor .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US

Title: System and Method for Scaleable Audio Transmission Over a Network

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TRANSMITTAL LETTER AND CERTIFICATE OF MAILING

Jun 9
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270

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-323-8979)
Lee & Hayes, PLLC
W. 201 North River Drive, Suite 430
Spokane, WA 99201

The following enumerated items accompany this transmittal letter and are being submitted for the matter identified in the above caption.

1. Transmittal letter including Certificate of Mailing
2. Response to Office Action Dated April 14, 1998
3. Return Post Card

Large Entity Status [x] Small Entity Status [ ]

The Commissioner is hereby authorized to charge payment of fees or credit overpayments to Deposit Account No. 50-0463 in connection with any patent application processing fees under 37 CFR 1.17; and any additional filing fees under 37 CFR 1.16 for the presentation of extra claims.

Date: May 11, 1998 By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34,656

CERTIFICATE OF MAILING

I hereby certify that the items listed above as enclosed are being deposited with the U.S. Postal Service as either first class mail, or Express Mail if the blank for Express Mail No. is completed below, in an envelope addressed to The Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below-indicated date. Any Express Mail No. has also been marked on the listed items.

Express Mail No. (if applicable) ____________________

Date: May 11, 1998 By: Helen M. Hare
Helen M. Hare

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO.
08/540,818	10/11/95	FERRIERE	P MS1069US

EXAMINER: JUNG, M

LM61/0617

LEE & HAYES, PLLC
W. 201 NORTH RIVER DRIVE SUITE 430
SPOKANE, WA 99201

ART UNIT	PAPER NUMBER
2731	11

DATE MAILED: 06/17/98

This is a communication from the examiner in charge of your application.
COMMISSIONER OF PATENTS AND TRADEMARKS

Min Jung
MIN JUNG
PATENT EXAMINER
GROUP ~~2600~~ 2731

NOTICE OF ALLOWABILITY

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to Amendment filed May 15, 1998.

☒ The allowed claim(s) is/are 5, 6, 8, 10-12, 9, 37-39, 13, 14, 19, 22, 20, 40, 21, 41, 23-28, 42-45 (renumbered as 1-28, respectively).

☒ The drawings filed on 10/11/95 are acceptable.

☐ Acknowledgement is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d).

☐ All ☐ Some* ☐ None of the CERTIFIED copies of the priority documents have been

☐ received.

☐ received in Application No. (Series Code/Serial Number) ______.

☐ received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

*Certified copies not received: ______.

☐ Acknowledgement is made of a claim for domestic priority under 35 U.S.C. § 119(e).

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ____.

☐ including changes required by the proposed drawing correction filed on ______, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal letter addressed to the Official Draftperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

☐ Notice of References Cited, PTO-892

☐ Information Disclosure Statement(s), PTO-1449, Paper No(s). ______

☐ Notice of Draftsperson's Patent Drawing Review, PTO-948

☐ Notice of Informal Patent Application, PTO-152

☒ Interview Summary, PTO-413

☒ Examiner's Amendment/Comment

☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material

☐ Examiner's Statement of Reasons for Allowance

PTOL-37 (Rev. 10/95)

★ U.S. GPO: 1996-404-496/405

11/C
D. Savoy
6/17/98

1. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with Mr. Lewis C. Lee on June 15, 1998.

2. The application has been amended as follows:

IN THE CLAIMS:

In claim 23, line 1, "as recited in claim 16" has been deleted.

In claim 23, line 2, "wherein" has been changed to ----comprising----.

3. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Min Jung whose telephone number is (703) 305-4363. The examiner can normally be reached on M-F from 9AM to 5PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chi Pham, can be reached on (703) 305-4378. The fax phone number for the organization where this application or proceeding is assigned is (703) 308-9051.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is (703) 305-3900.

M.JUNG

June 15, 1998

Min Jung

MIN JUNG
PATENT EXAMINER
GROUP ~~2600~~ 2731

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKETT NO.
08/540,818	10/11/95	FERRIERE P	MS1069US

LM61/0617

LEE & HAYES, PLLC
W. 201 NORTH RIVER DRIVE SUITE 430
SPOKANE, WA 99201

EXAMINER	
JUNG,M	
ART UNIT	**PAPER NUMBER**
2731	11

DATE MAILED: 06/17/98

EXAMINER INTERVIEW SUMMARY RECORD

All participants (applicant, applicant's representative, PTO personnel):

(1) Lewis C. Lee (3) ______

(2) Min Jung (4) ______

Date of Interview June 15, 1998

Type: ☒ Telephonic ☐ Personal (copy is given to ☐ applicant ☐ applicant's representative).

Exhibit shown or demonstration conducted: ☐ Yes ☐ No. If yes, brief description: ______

Agreement ☒ was reached with respect to some or all of the claims in question. ☐ was not reached.

Claims discussed: 23

Identification of prior art discussed: ______

Description of the general nature of what was agreed to if an agreement was reached, or any other comments: It was noted that claim 23 depends from a cancelled claim. It was agreed to make changes in claim language to make it in an independant form. See the attached examiner's amendment.

(A fuller description, if necessary, and a copy of the amendments, if available, which the examiner agreed would render the claims allowable must be attached. Also, where no copy of the amendments which would render the claims allowable is available, a summary thereof must be attached.)

☐ 1. It is not necessary for applicant to provide a separate record of the substance of the interview.

Unless the paragraph below has been checked to indicate to the contrary, A FORMAL WRITTEN RESPONSE TO THE LAST OFFICE ACTION IS NOT WAIVED AND MUST INCLUDE THE SUBSTANCE OF THE INTERVIEW (e.g., items 1-7 on the reverse side of this form). If a response to the last Office action has already been filed, then applicant is given one month from this interview date to provide a statement of the substance of the interview.

☐ 2. Since the examiner's interview summary above (including any attachments) reflects a complete response to each of the objections, rejections and requirements that may be present in the last Office action, and since the claims are now allowable, this completed form is considered to fulfill the response requirements of the last Office action. Applicant is not relieved from providing a separate record of the substance of the interview unless box 1 above is also checked.

Min Jung
Examiner's Signature

PTOL-413 (REV. 2 -93)

ORIGINAL FOR INSERTION IN RIGHT HAND FLAP OF FILE WRAPPER

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

LM61/0617

LEE & HAYES, PLLC
W. 201 NORTH RIVER DRIVE SUITE 430
SPOKANE, WA 99201

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT		DATE MAILED
08/540,818	10/11/95	028	JUNG, M	2731	06/17/98

First Named Applicant: FERRIERE, PHILIPPE

TITLE OF INVENTION: SYSTEM AND METHOD FOR SCALEABLE STREAMED AUDIO TRANSMISSION OVER A NETWORK

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 MS1069US	370-465.000	F41	UTILITY	NO	$1320.00	09/17/98

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above.
If the SMALL ENTITY is shown as YES, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown above and notify the Patent and Trademark Office of the change in status, or
B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statement of Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B-Issue Fee Transmittal should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B Issue Fee Transmittal should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "4b" of Part B-Issue Fee Transmittal should be completed and an extra copy of the form should be submitted.

III. All communications regarding this application must give application number and batch number. Please direct all communications prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: Utility patents issuing on applications filed on or after Dec. 12, 1980 may require payment of maintenance fees. It is patentee's responsibility to ensure timely payment of maintenance fees when due.

PATENT AND TRADEMARK OFFICE COPY

PTOL-85 (REV. 10-96) Approved for use through 06/30/99. (0651-0033)

*U.S. GPO: 1998-437-639/80023

PART B—ISSUE FEE TRANSMITTAL

Complete and mail this form, together with applicable fees, to: **Box ISSUE FEE**
Assistant Commissioner for Patents
Washington, D.C. 20231

B.H.

D.S.

MAILING INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE. Blocks 1 through 4 should be completed where appropriate. All further correspondence including the Issue Fee Receipt, the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

CURRENT CORRESPONDENCE ADDRESS (Note: Legibly mark-up with any corrections or use Block 1)

LEE & HAYES, PLLC
W. 201 NORTH RIVER DRIVE SUITE 430
SPOKANE, WA 99201

RECEIVED 17
Publishing Division
AUG 3 1 1998

11

Note: The certificate of mailing below can only be used for domestic mailings of the Issue Fee Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing.

Certificate of Mailing

I hereby certify that this Issue Fee Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Box Issue Fee address above on the date indicated below.

Helen M. Hare (Depositor's name)

Helen M. Hare (Signature)

August 27 1998 (Date)

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT		DATE MAILED
08/540,818	10/11/95	028	JUNG, M	2731	06/17/98

First Named Applicant: FERRIERE, PHILIPPE

TITLE OF INVENTION: SYSTEM AND METHOD FOR SCALEABLE STREAMED AUDIO TRANSMISSION OVER A NETWORK

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 MS1069US	370-465.000	F41	UTILITY	NO	$1320.00	09/17/98

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363). Use of PTO form(s) and Customer Number are recommended, but not required.

☐ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.

☐ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47) attached.

2. For printing on the patent front page, list (1) the names of up to 3 registered patent attorneys or agents OR, alternatively, (2) the name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.

1 Lee & Hayes, PLLC
2 ______
3 ______

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)
PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. Inclusion of assignee data is only appropiate when an assignment has been previously submitted to the PTO or is being submitted under separate cover. Completion of this form is NOT a subsititue for filing an assignment.

(A) NAME OF ASSIGNEE Microsoft Corporation

(B) RESIDENCE: (CITY & STATE OR COUNTRY) Redmond, Washington

Please check the appropriate assignee category indicated below (will not be printed on the patent)

☐ individual ☒ corporation or other private group entity ☐ government

4a. The following fees are enclosed (make check payable to Commissioner of Patents and Trademarks):

☐ Issue Fee
☐ Advance Order - # of Copies ______

4b. The following fees or deficiency in these fees should be charged to:
DEPOSIT ACCOUNT NUMBER 50-0463
(ENCLOSE AN EXTRA COPY OF THIS FORM)

☒ Issue Fee
☒ Advance Order - # of Copies 8

The COMMISSIONER OF PATENTS AND TRADEMARKS IS requested to apply the Issue Fee to the application identified above.

(Authorized Signature)
Lewis C. Lee #34,656 [signature]

(Date) Aug. 27, 1998

NOTE; The Issue Fee will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the Patent and Trademark Office.

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending on the needs of the individual case. Any comments on the amount of time required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, D.C. 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND FEES AND THIS FORM TO: Box Issue Fee, Assistant Commissioner for Patents, Washington D.C. 20231

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

09/11/1998 CASHBY 00000195 500463 08540818

01 FC:142 1320.00 CH
02 FC:561 24.00 CH

TRANSMIT THIS FORM WITH FEE

PTOL-85B (REV.10-96) Approved for use through 06/30/99. OMB 0651-0033

Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

PTO UTILITY GRANT

Paper Number 12

The Commissioner of Patents and Trademarks

Has received an application for a patent for a new and useful invention. The title and description of the invention are enclosed. The requirements of law have been complied with, and it has been determined that a patent on the invention shall be granted under the law.

Therefore, this

United States Patent

Grants to the person(s) having title to this patent the right to exclude others from making, using, offering for sale, or selling the invention throughout the United States of America or importing the invention into the United States of America for the term set forth below, subject to the payment of maintenance fees as provided by law.

If this application was filed prior to June 8, 1995, the term of this patent is the longer of seventeen years from the date of grant of this patent or twenty years from the earliest effective U.S. filing date of the application, subject to any statutory extension.

If this application was filed on or after June 8, 1995, the term of this patent is twenty years from the U.S. filing date, subject to an statutory extension. If the application contains a specific reference to an earlier filed application or applications under 35 U.S.C. 120, 121 or 365(c), the term of the patent is twenty years from the date on which the earliest application was filed, subject to any statutory extension.

The United States of America

PATENT AND TRADEMARK OFFICE · UNITED STATES OF AMERICA

Bruce Lehman

Commissioner of Patents and Trademarks

Sandra L Morton

Attest

Form PTO-1584 (Rev. 2/97)

Please type a plus sign (+) inside this box → +

PTO/SB/123 (8-96)
Approved for use through 6/30/99. OMB 0651-0035
Patent and Trademark Office: U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

CHANGE OF CORRESPONDENCE ADDRESS
Patent

Address to:
Assistant Commissioner for Patents
Washington, D.C. 20231

Patent Number	5,835,495
Issue Date	11/10/98
Application Number	08/540,818
Filing Date	10/11/95
First Named Inventor	Ferriere

Please change the Correspondence Address for the above-identified patent to:

[X] Customer Number 22801 → *Place Customer Number Bar Code Label here*

OR *Type Customer Number here*

[ ]

Firm *or* Individual Name	Lee & Hayes, PLLC
Address	
Address	
City	State	ZIP
Country	
Telephone	Fax

This form cannot be used to change the data associated with a Customer Number. To change the data associated with an existing Customer Number use "Request for Customer Number Data Change" (PTO/SB/124).

This form will not affect any "fee address" provided for the above-identified patent. To change a "fee address" use the "Fee Address Indication Form" (PTO/SB/47).

I am the :

[ ] Patentee.

[ ] Assignee of record of the entire interest.
Certificate under 37 CFR 3.73(b) is enclosed.

[X] Attorney or agent of record .

Typed or Printed Name	Lewis C. Lee
Signature	Lewis C. Lee
Date	10/6/99

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending upon the needs of the individual case. Any comments on the amount of time you are required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, DC 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Assistant Commissioner for Patents, Washington, DC 20231.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

#13
CD

Patent No 5,835,495
Issue Date 11/10/98
Inventorship .... Ferriere
Applicant .... Microsoft Corporation
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

TRANSMITTAL LETTER AND CERTIFICATE OF MAILING

CERTIFICATE
OCT 2 1 1999
OF CORRECTION

To: Commissioner of Patents and Trademarks,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-323-8979)
Lee & Hayes, PLLC
421 W. Riverside Avenue, Suite 500
Spokane, WA 99201

The following enumerated items accompany this transmittal letter and are being submitted for the matter identified in the above caption.

1. Transmittal letter including Certificate of Mailing
2. Return Post Card
3. Request for Certificate of Correction
4. Certificate of Correction
5. Change of Correspondence Address

APPROVED
MAR 21 2000
FOR THE COMMISSIONER OF PAT & TM

Large Entity Status [x] Small Entity Status [ ]

Applicant hereby requests an extension of time in any case such an extension is necessary. The fee should be charged to the Deposit Account indicated below.

The Commissioner is hereby authorized to charge payment of fees or credit overpayments to Deposit Account No. 50-0463 in connection with any patent application filing fees under 37 CFR 1.16, and any processing fees, including any necessary fees under 37 CFR 1.17 or 37 CFR 1.20.

Date: 10/6/99 By: [signature]
Lewis C. Lee
Reg. No. 34,656

CERTIFICATE OF MAILING

I hereby certify that the items listed above as enclosed are being deposited with the U.S. Postal Service as first class mail in an envelope addressed to The Commissioner of Patents and Trademarks, Washington, D.C. 20231, on the below-indicated date

Date: Oct. 6, 1999 By: Helen M. Hare
Helen M. Hare

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Patent No. 5,835,495
Dated ....Nov. 10, 1998
Inventor..Ferriere
Group Art Unit 2731
Examiner .... Min Jung
Attorney's Docket No. MS1-069US
Title: System and Method for Scaleable Streamed Audio Transmission over a Network

REQUEST FOR CERTIFICATE OF CORRECTION

References -- See Attached Form PTO/SB/44, one page

To: Certificate of Correction Branch
Assistant Commissioner for Patents,
Washington, D.C. 20231

From: Lewis C. Lee (Tel. 509-324-9256; Fax 509-323-8979)
Lee & Hayes, PLLC
421 W. Riverside Avenue, Suite 500
Spokane, WA 99201

Applicant submits herewith a request for the issuance of a Certificate of Correction for U.S. Patent No. 5,835,495 pursuant to 37 C.F.R. 1.323, *Certificate of correction of applicant's mistake*. The appropriate fee for providing the correction, as set forth in 37 C.F.R. 1.20(a), is included herewith.

Applicant respectfully asserts that the mistakes are typographical in nature and of minor character. Further, the proposed corrections do not involve changes which would constitute new matter or require reexamination.

Respectfully Submitted,

Date: 10/6/99

By: Lewis C. Lee
Lewis C. Lee
Reg. No. 34656
(509) 324-9256

NOTICE RE: CERTIFICATES OF CORRECTION

DATE: **March 7, 2000**

Paper No. **14**

TO: Supervisor, Art Unit **2731**

SUBJECT: Certificate of Correction Request in Patent No.: **5,835,495**

A response to the following question(s) is requested with respect to the accompanying request for a certificate of correction.

- [X] 1. Would the change(s) requested, under 37 C.F.R. 1.323 constitute new matter or require reexamination of the application? **SEE CHANGES TO FIG. 4 AND COL. 14, LINE 45.**
- [ ] 2. Would the change(s) requested under 37 C.F.R 1.323 materially affect the scope or meaning of the claims allowed by the examiner in the patent?
- [ ] 3. Applicant disagrees with change(s) initialed and dated by Examiner in lieu of an Examiner's Amendment. Should the change request be granted?
- [ ] 4. With respect to the change(s) requested, correcting Office errors, should the patent read as shown in the certificate of correction?.
- [ ] 5. If the amendments filed ____________ had been considered by the Examiner, would the amendment have been entered?

PLEASE RESPOND WITHIN 7 DAYS AND RETURN THE FILE TO PALM LOCATION. 7580, CERTIFICATES OF CORRECTION BR, PARK 3 -918, Thank you

Chantae Dessau

Legal Instrument Examiner

PLEASE CHECK THE BOX(ES) BELOW CORRESPONDING TO THE BOXES CHECKED FOR QUESTION(S) ABOVE AND RETURN FILE TO: PALM LOCATION 7580, CERT. OF CORREC. BR., PK 3 - 918

DATE:

The decision regarding the change(s) requested in the certificate of correction is shown below.

[ ] 1.YES	[✓] NO	[ ] Comments below
[ ] 2.YES	[ ] NO	[ ] Comments below
[ ] 3.YES	[ ] NO	[ ] Comments below
[ ] 4.YES	[ ] NO	[ ] Comments below
[ ] 5.YES	[ ] NO	[ ] Comments below

[ ] Comments ____________

Chau T. Nguyen 3/15/00

Supervisor

2739

Art Unit

PTOL-306 (REV. 10/87)

U.S. DEPARTMENT OF COMMERCE Patent and Trademark Office

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
ASSISTANT SECRETARY AND COMMISSIONER
OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

CHANGE OF ADDRESS/POWER OF ATTORNEY

FILE LOCATION 9200 SERIAL NUMBER 08540818 PATENT NUMBER 5835495

THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER # 22801

THE PRACTITIONERS OF RECORD HAVE BEEN CHANGED TO CUSTOMER # 22801

ON 08/29/01 THE ADDRESS OF RECORD FOR CUSTOMER NUMBER 22801 IS:

LEE & HAYES PLLC
421 W RIVERSIDE AVENUE SUITE 500
SPOKANE WA 99201

AND THE PRACTITIONERS OF RECORD FOR CUSTOMER NUMBER 22801 ARE:

34521	34618	34656	37773	37954	38318	38605	39241	39384	40559
42905	42973	44302	44421	44453					

PTO INSTRUCTIONS: PLEASE TAKE THE FOLLOWING ACTION WHEN THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER NUMBER: RECORD, ON THE NEXT AVAILABLE CONTENTS LINE OF THE FILE JACKET, 'ADDRESS CHANGE TO CUSTOMER NUMBER'. LINE THROUGH THE OLD ADDRESS ON THE FILE JACKET LABEL AND ENTER ONLY THE 'CUSTOMER NUMBER' AS THE NEW ADDRESS. FILE THIS LETTER IN THE FILE JACKET. WHEN ABOVE CHANGES ARE ONLY TO FEE ADDRESS AND/OR PRACTITIONERS OF RECORD, FILE LETTER IN THE FILE JACKET.
THIS FILE IS ASSIGNED TO GAU 2731.

PTO-FMD
TALBOT-1/97

Form PTO 1130
(REV 2/94)

PACE DATA ENTRY CODING SHEET

U.S. DEPARTMENT OF COMMERCE
Patent and Trademark Office

1ST EXAMINER	336	DATE	12/5/95
2ND EXAMINER	401	DATE	3/11/96

APPLICATION NUMBER	TYPE APPL	FILING DATE (MONTH DAY YEAR)	SPECIAL HANDLING	GROUP ART UNIT	CLASS	SHEETS OF DRAWING
08/540818	1	10 11 95	0	2603	370	006

TOTAL CLAIMS	INDEPENDENT CLAIMS	SMALL ENTITY?	FILING FEE	FOREIGN LICENSE	ATTORNEY DOCKET NUMBER
036	010	0	1778	Y	MS106948

CONTINUITY DATA

CONT CODE	STATUS CODE	PARENT APPLICATION SERIAL NUMBER	PCT APPLICATION SERIAL NUMBER	PARENT PATENT NUMBER	PARENT FILING DATE (MONTH DAY YEAR)
			PCT/ /		
			PCT/ /		
			PCT/ /		
			PCT/ /		
			PCT/ /		

PCT/FOREIGN APPLICATION DATA

FOREIGN PRIORITY CLAIMED	COUNTRY CODE	PCT/FOREIGN APPLICATION SERIAL NUMBER	FOREIGN FILING DATE (MONTH DAY YEAR)

PATENT APPLICATION FEE DETERMINATION RECORD Effective October 1, 1995	Application or Docket Number 540818

CLAIMS AS FILED - PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	SMALL ENTITY RATE	SMALL ENTITY FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY FEE
BASIC FEE				375.00	OR		750.00
TOTAL CLAIMS	36 minus 20 =	* 16	x$11=		OR	x$22=	352
INDEPENDENT CLAIMS	10 minus 3 =	* 7	x39=		OR	x78=	546
MULTIPLE DEPENDENT CLAIM PRESENT			+125=		OR	+250=	
			TOTAL		OR	TOTAL	1648

* If the difference in column 1 is less than zero, enter "0" in column 2

CLAIMS AS AMENDED - PART II

AMENDMENT A

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	* 39	Minus	** 36	= 3	x$11=		OR	x$22=	66.00
Independent	* 17	Minus	*** 10	= 7	x39=		OR	8200 x78=	546.00
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+125=		OR	+250=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	Paid

AMENDMENT B

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	* 28	Minus	** 39	= —	x$11=		OR	x$22=	
Independent	* 12	Minus	*** 17	= —	x39=		OR	x78=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+125=		OR	+250=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x39=		OR	x78=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+125=		OR	+250=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20."
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3."
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

FORM PTO-875
(Rev. 10/95)

Patent and Trademark Office, U.S. DEPARTMENT OF COMMERCE

STAPLE AREA

U.S. GOVERNMENT PRINTING OFFICE: 1997-430-220

PATENT NUMBER	ORIGINAL CLASSIFICATION	
	CLASS: 370	SUBCLASS: 465

APPLICATION SERIAL NUMBER
08/540,818

APPLICANT'S NAME (PLEASE PRINT)
Ferriere

IF REISSUE, ORIGINAL PATENT NUMBER

CROSS REFERENCE(S)

CLASS	SUBCLASS (ONE SUBCLASS PER BLOCK)			
370	521			
375	222			
379	68			

INTERNATIONAL CLASSIFICATION 6

H	0	4	J	3/18

GROUP ART UNIT: 2731

ASSISTANT EXAMINER (PLEASE STAMP OR PRINT FULL NAME)

PRIMARY EXAMINER (PLEASE STAMP OR PRINT FULL NAME)
Min Jung

PTO 270 (REV. 5-91)

ISSUE CLASSIFICATION SLIP

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE

Staple Issue Slip Here

POSITION		ID NO.	DATE
CLASSIFIER		20	11/30
EXAMINER		2310	12/5/95
TYPIST		452	3-28-96
VERIFIER		VN00	
CORPS CORR.			
SPEC. HAND		401	3/11/96
FILE MAINT.		18	12-7
DRAFTING			

INDEX OF CLAIMS

Claim: Final	Claim: Original	Date: 9/18/97	4/9/98	6/15/98					
	1	✓							
	2								
	3								
	4								
1	5		#	=					
2	6		#	=					
	7								
3	8		=	=					
7	9		=	=					
4	10		=	=					
5	11		=	=					
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15	20		=	=					
17	21		=	=					
14	22		=	=					
19	23		=	=					
20	24		=	=					
21	25		=	=					
22	26		=	=					
23	27	✓	=	=					
24	28	✓	=	=					
	29	+	+						
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	32								
	33								
	34								
	35								
	36	+	+						
8	37		=	=					
9	38		=	=					
10	39		=	=					
16	40		=	=					
18	41		=	=					
25	42		=	=					
26	43		=	=					
27	44		=	=					
28	45		=	=					
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Claim: Final	Claim: Original	Date							
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	100								

SYMBOLS

- ✓ .... Rejected
- = ... Allowed
- \- (Through numberal) Canceled
- \+ .... Restricted
- N .... Non-elected
- I .... Interference
- A .... Appeal
- O .... Objected

(LEFT INSIDE)

[redacted]

SEARCHED

Class	Sub.	Date	Exmr.
370	252 465 486 521	9/16/97 9/17/97	mJ
375	222		
379	68 88 89		
updated		4/9/98	mJ
370	477	6/12/98	mJ
375	241 242 245		

INTERFERENCE SEARCHED

Class	Sub.	Date	Exmr.
370	252 465 486 521 477	6/15/98	mJ
375	222 241 242 245		
379	68 88 89		

SEARCH NOTES

	Date	Exmr.
Consulted Fan Tsang (Cl. 379) Consulted Stephen Chin (Cl. 375)	9/16/97	mJ.
APS.	9/16/97 9/17/97	mJ

(RIGHT OUTSIDE)