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(54) A METHOD AND SYSTEM FOR REALIZING THE VOICE COMPENSATION IN THE MOBILE COMMUNICATION NETWORK

VERFAHREN UND SYSTEM ZUR REALISIERUNG DER SPRACHKOMPENSATION IM
MOBILKOMMUNIKATIONSNETZ

PROCÉDÉ ET SYSTÈME PERMETTANT DE RÉALISER UNE COMPENSATION VOCALE DANS
UN RÉSEAU DE COMMUNICATION MOBILE

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(73) Proprietor: **ZTE Corporation**
Shenzhen city, Guangdong 518057 (CN)

(72) Inventors:

- **LU, Donghua**
Shenzhen City
Guangdong 518057 (CN)
- **RUAN, Wei**
Shenzhen City
Guangdong 518057 (CN)
- **CAO, Jian**
Shenzhen City
Guangdong 518057 (CN)
- **LOU, Hongwei**
Shenzhen City
Guangdong 518057 (CN)

• **ZHANG, Wanchun**
Shenzhen City
Guangdong 518057 (CN)

(74) Representative: **Page Hargrave**
Whitefriars
Lewins Mead
Bristol BS1 2NT (GB)

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Description

Technical Field

[0001] The present invention relates to voice compensation technology, and more specifically, to a method and system for voice compensation when the device at network side does not use or partially use a vocoder.

Background

[0002] In a mobile communication system, vocoder at network side mainly has the following two important roles: in the uplink, after the user equipment (UE) sends the compressively encoded voice to the network side, the vocoder at the network side decodes the received compressed voice to make the voice suitable to be transmitted in the network; in the downlink, the vocoder at the network side compressively encodes the voice stream transmitted in the network to make the voice suitable to be transmitted in the radio link.

[0003] Taking CDMA2000 (Code Division Multiple Access 2000) as an example, there are three methods for voice encoding and decoding in CDMA2000: EVRC (Enhanced Variable Rate Coder), QCELP-13k (Qualcomm Code Excited Linear Predictive Coding-Qualcomm-13k) and QCELP-8k (Qualcomm Code Excited Linear Predictive Coding-Qualcomm-8k). Wherein, EVRC is the most widely used mainstream encoding and decoding format. In a typical call between MS1 (Mobile Station) and MS2, they use the same voice encoding and decoding method (such as EVRC). The voice of the MS1 subscriber arrives at the ears of the MS2 subscriber through the following way: Firstly, MS1 sends the encoded EVRC compressed voice frame to network side 1 through the radio uplink, and network side 1 uses vocoder 1 to decode the received EVRC voice frames and transform them into circuit PCM (Pulse Coded Modulation) code stream, and then perform circuit switching. After network side 2 receives the circuit-switched PCM code stream from network side 1, it uses vocoder 2 to transform the PCM code stream to EVRC compressed voice frames and sends them to MS2 through radio downlink.

[0004] The voice encoding and decoding by the vocoder is lossy compression, each encoding and decoding will decrease the voice quality. Still taking the call between MS1 and MS2 as an example, since MS1 and MS2 use the same encoding and decoding format, two voice encoding and decoding processes can be avoided at the network side if the encoding and decoding of EVRC compressed voice frames added at the network side can be avoided, and the process that the voice of MS1 subscriber arrives at the ears of MS2 subscriber is as follows: firstly, MS1 sends the encoded EVRC compressed voice frames to network side 1 through radio uplink, and network side 1 directly switches the received EVRC voice frames and sends them to network side 2; network side 2 receives the switched EVRC compressed voice frames

and sends them to MS2 through radio downlink.

[0005] From this example, it can be seen that, since two voice encoding and decoding processes which will damage the voice quality are avoided at the network side, not only the voice quality is evidently improved, but also the vocoder resource at the network side is saved, thus reducing the delay for transmitting and processing the voice. During the initial development stage of mobile communication system, the voice calls in system are mainly between mobile subscribers and fixed subscribers, therefore, the above influence is not evident. Traffic statistics shows that, the calls among the mobile subscribers have already taken the leading position, and thus the original vocoder configuration will not only add the device cost but also affect the system performance. Therefore, how to improve the network structure and strategy of the vocoder configuration management becomes the research hotspot.

[0006] Patent application WO 02/054744 A1 discloses a network element and a method for enhancing the quality of digitised analogue signals transmitted in parameterised coded form via a digital network. According to this patent application, in order to enable an enhancement of the quality of the digitised analogue signals on network side, the network element comprises means for extracting signals from and insert signals into the network, first processing means for processing the extracted parameters in the parameter domain with functions suitable to enhance the quality of the digitised analogue signals and second processing means for processing the extracted parameters in the linear domain with functions suitable to enhance the quality of the digitised analogue signals.

[0007] With the development of all-IP mobile communication technology, supporting the traditional voice service and packet data service with lower cost and more flexible method is the main driving force of the development of all-IP mobile communication technology. When supporting traditional voice service, the all-IP mobile communication network has the problem of how to simultaneously support several kinds of vocoders with lower cost, that is, the problem of supporting so-called TrFO (Transcoder Free Operation) and RTO (Remote Transcoder Operation).

[0008] TrFO means making the network negotiate the encoding and decoding type and the mode of the vocoder before call establishment through some out-band negotiation mechanism. Through negotiation, calls among the mobile subscribers do not need to pass through the vocoder at network side, thus improving the voice quality, saving expensive vocoder resource and the followed power consumption.

[0009] RTO is a special case of TrFO, and since the encoding and decoding methods of both sides of communication are not consistent during out-band negotiation, it is necessary that there is a vocoder at network side to transform the code type of one side of communication into the code type of the other side. The main difference between RTO and TDM circuit transmission net-

works is that: in TDM network, there are two encoding and decoding transformations at network side, while only one in RTO network. Taking RTO as an example here, MS1 uses EVRC while MS2 uses QCELP-13k, during the call between MS1 and MS2, the voice of MS1 subscriber arrives at the ears of MS2 subscriber through the following process: firstly, MS1 sends the encoded EVRC compressed voice frames to network side 1 through radio uplink; network side 1 directly switches the received EVRC voice frames and sends them to network side 2; network side 2 receives the switched EVRC compressed voice frames and transforms them into QCELP-13k compressed voice frames by a vocoder, and sends them to MS2 through radio downlink.

[0010] Taking CDMA2000 LMSD (Legacy Mobile Station Domain) as an example, out-band negotiation of TrFO is implemented through signaling negotiation between access network and MSCe. Since CDMA2000 LMSD applies IP switching technology, network side can directly make the compressed voice data encoded by the UE be RTP (Real-Time Transport Protocol) packets and transfer them through IP network, and there is no need to transform encoding and decoding scheme to PCM and to be transmit through TDM circuit.

[0011] Taking EVRC as an example, as inter alia disclosed in 3rd Generation Partnership Project 2, "3GPP2", "Enhanced Variable Rate Codec, Speech Service Option 3 and 68 for Wideband Spread Spectrum Digital Systems", 3GPP2 C.S0014-B, Version 1.0, May 2006, the maximal transmission rate of EVRC is 8kbps (the transmission rate of full-rate frame), and EVRC also has lots of half-rate frames and 1/8-rate frames. Statistical analysis shows that, in EVRC call, the proportion of full-rate frames is averagely around 30%, and its transmission rate is 22 bytes/20ms frame; the proportion of half-rate frames is averagely around 30%, and its transmission rate is 10 bytes/20ms frame; the proportion of 1/8-rate frames is averagely around 40%, and its transmission rate is 2 bytes/20ms frame. In addition, since RTP transmission supports multi-frame packing function, EVRC can be packed to be transmitted in the network to save the overhead of IP header. Taking packing 3 EVRC frames into a RTP message for example, the average transmission rate of EVRC in the network is 11.7kbps with the overhead of IP header. While the transmission rate of one-path voice PCM code stream in network is 64kbps in the previous TDM circuit transmission, thus the bandwidth used to transmit compressed voice in all-IP saves about $(1-11.7/64)=81.7\%$ of PCM code stream bandwidth in TDM circuit mode. This example illustrates that, TrFO can save a lot of network bandwidth.

[0012] However, TrFO has some problems in the practical applications. For example, suppose MS1 and MS2 are in a TrFO call, if the quality of radio link is not good enough, there is possibility that network side 1 cannot correctly receive and analyze the content of some frames sent by MS 1 to network side 1 through uplink, that is, frame error in the air. These frames that cannot be ana-

lyzed can be smoothly processed by the vocoder at the network side in TDM circuit transmission network, while in TrFO technology, since there is no participation of vocoder, network side 1 only can fill these frame errors with to-be-compensated frames specified in protocol (for example, in EVRC, half-rate or full-rate frames with all bits being "0" are defined as to-be-compensated frames, and so on) and switch them to network side 2 which sends these to-be-compensated frames specified by protocol to MS2. Meanwhile, because of the characteristic of IP network transmission, there is possibility of frame loss or frame jittering during the process of the voice frames at network side 1 arriving at network side 2 through network, at this time, if network side 2 does not receive frames from network side 1 in the specified time, it will fill the frame errors with to-be-compensated frames according to the protocol and send them to MS2. After these to-be-compensated frames due to the quality of radio link and the quality of network transmission arrive at MS2, there would be no problem if MS2 performs voice compensation for these frames; however, a lot of experiments show that, most UEs will not perform voice compensation for these to-be-compensated frames. Therefore, these to-be-compensated frames will have a big influence on the whole voice quality of TrFO.

[0013] RTO technology also has the same problem. Since RTO call uses vocoder at network side, suppose MS1 and MS2 are in RTO call, some of the frames sent by MS1 to network side 1 through uplink may have errors if the quality of radio link is not good enough, network side 1 can use the vocoder at network side to perform voice compensation for these frame errors. However, when the compensated voice frames arrives at network side 2, there is still possibility of frame loss and frame jittering due to the problem of network transmission quality, at this time, network side 2 will fill the frame errors with to-be-compensated frames specified in protocol and send them to MS2. Therefore, these to-be-compensated frames will have big influence on the whole voice quality of RTO if voice compensation cannot be performed effectively for these frames by MS2.

[0014] To sum up, when the environment of radio link is good and the quality of network transmission is ideal, TrFO and RTO reducing the times of encoding and decoding by the vocoder at network side will improve the voice quality, but they cannot use the vocoder at network side for voice compensation as the original circuit-switched mobile communication system does when the environment of radio link is not good and the quality of network transmission is poor. At this time, voice compensation totally relies on the vocoder at the UE side. To date, the UEs produced by various manufacturers in the market have different solutions regarding to whether it needs to compensate the received to-be-compensated voice frames or not, and thus the voice qualities of TrFO and RTO severely rely on the compensation performance of the vocoder in the UE and on whether the vocoder compensates all kinds of to-be-compensated voice

frames or not, which will significantly affect the whole voice quality of TrFO and RTO.

[0015] Practice proves that, if one of consecutive full-rate frames is damaged or lost, what the UE receives is to-be-compensated frames in the case of TrFO or RTO. The voice quality when the UE processes the to-be-compensated frames is evidently worse than the voice quality in the case that there is vocoder at network side in TDM circuit transmission network, there are phenomena of words swallowing, trilling and discontinuous voice in the former case. For the UEs having different vocoders, the degrees of how good or poor the voice quality is are not the same.

Summary of the Invention

[0016] The present invention offers a method and system for realizing voice compensation in a mobile communication network to solve the above disadvantages and to approximately compensate the voice in the condition that the transmission quality is poor and the device at network side does not use or partially uses vocoder, thus improving the quality of whole voice.

[0017] The technique scheme applied in the present invention is:

[0018] A method for realizing voice compensation in mobile communication network, as set forth in independent claim 1.

[0019] The present invention also offers a system for realizing voice compensation in mobile communication network, as set forth in independent claim 3.

[0020] Preferred embodiments are set forth in the dependent claims.

[0021] The system and method of the present invention can effectively solve the problem that the voice quality makes the ears uncomfortable, including discontinuous phenomenon, trilling and swallowing words during the call, due to the bad radio link environment or poor network transmission quality when there is no vocoder at the network side participates or vocoder at the network side partially participates the call. The scheme of the present invention implements voice compensation in the device at the network side to effectively reduce the dependency of the call on the user terminal and the vocoder's performance, thus satisfying all kinds of voice quality requirements by the user terminals.

Brief Description of Drawings

[0022]

FIG. 1 is a flow chart of specifically implementing the method for voice compensation in accordance with the present invention;

FIG. 2 is an illustration of the system for specifically implementing voice compensation in accordance with the present invention;

FIG. 3 is a flow chart in accordance with the embod-

iment 1 of the present invention;

FIG 4 is a flow chart in accordance with the embodiment 2 of the present invention;

FIG 5 is a flow chart in accordance with the embodiment 3 of the present invention.

Preferred Embodiments of the Present Invention

[0023] The present invention will be described in further detail with reference to the accompanying figures and embodiments.

[0024] The main idea of the present invention is: during the call, full-rate frames and half-rate frames gives the greatest contribution to the voice, if there are full-rate frames or half-rate frames damaged or lost, the voice quality will be easily affected. A lot of experiments show that, the loss or damage of one or more full-rate frames will cause discontinuous voice, swallowing words, especially in the condition of consecutive full-half-rate frames; and the loss or damage of one or more half-rate frames will cause trilling especially in the condition of consecutive full-half-rate frames, thus making the ears uncomfortable, and the degree of uncomfortable feeling relies on the encoding and decoding performance of the vocoder of the UE. Therefore, the objective of the present invention is to mainly compensate full-rate frames or half-rate ones.

[0025] The present invention offers a method for realizing voice compensation in mobile communication network to apply in the condition that the radio environment is bad or transmission quality is poor, and the network side does not use or partially uses vocoder (such as TrFO or RTO), as shown in FIG. 1, and the method comprises the following steps:

Step 1: The device at the network side judges the forward frame received from the network side or to be sent and processed is an invalid frame or not at each time of processing the forward voice frames; or the network side judges the backward frame received from the UE or to be sent and processed is an invalid frame or not at each time of processing the backward voice frames;

If it is invalid frame, the process proceeds to step 2; Otherwise, normally process and output the voice frames.

Said "invalid frames" means the following kinds of frames:

Blank frames, deleted frames, or other undefined-rate frames in protocol;

Or frames not received at the specified time of processing the frames (such as loss frames or frames arrived later due to jittering);

Or frames needed to be compensated after being received by the vocoder specified in the protocol.

Step 2: the device at the network side judges whether the voice compensation should be performed for the invalid frame. The basis of judgment is whether the invalid frame is the frame in the condition of non-1/8-rate :

If the invalid frame is a frame in the condition of non-1/8-rate, the invalid frame will have a relatively big influence on the voice quality, the process proceeds to step 3;

If the invalid frame is a frame in the condition of 1/8-rate, the invalid frame will have little influence on the voice quality and does not need to be compensated, the invalid frame is normally processed and output.

[0026] The method for judging whether the invalid frame is a frame in the condition of non-1/8-rate is as follows:

The device at the network side judges whether the last valid frame is a 1/8-rate frame or not, if the last valid frame is a non-1/8-rate frame, the invalid frame is a frame in the condition of non-1/8-rate; otherwise, the invalid frame is a frame in the condition of 1/8-rate;

If the device at the network side judges each forward voice frame in step 1, it judges the last valid frame before the forward voice frame in this step; if the device at the network side judges each backward voice frame in step 1, it judges the last valid frame before the backward voice frame in this step.

[0027] Said "valid frame" means that the frame can be normally encoded and decoded by the vocoder during the call, that is, those frames other than the invalid frames are called valid frames.

[0028] Said "last valid frame" means the valid frame received or to be sent at the last time of processing frames; if the frame received or to be sent at the last time of processing frames is an invalid frame, it means the valid frame received or to be sent at the time just before the last one of processing frames, and so on.

[0029] Step 3: the device at the network side judges whether the frame distance between the invalid frame and its last valid frame is less than or equals to the threshold of voice compensation:

If yes, the process proceeds to step 4;

Otherwise, there is no voice compensation, and the invalid frame is normally processed and output.

[0030] Said compensation threshold depends on the compensation effect and performance of the mobile communication system, the compensation threshold to achieve the optimal compensation effect can be obtained according to the voice quality by comparing the results of several experiments; For example, the compensation

threshold is set as 6, 6 consecutive invalid frames are compensated; while if the compensation threshold is set as 2, only two invalid frames are compensated, and the third one of the consecutive frames will not be compensated.

[0031] For the "frame distance", for example, the frame distance between frame A and frame B means the number of frames between frame A and frame B plus 1 for a group of sequentially arrived frames. For example, for a group of sequentially arrived frames, frame a, frame b, frame c, frame d, the frame distance between frame a and frame d is 3.

[0032] Step 4: the device at the network side performs voice frame compensation for the invalid frame, uses the compensated voice frame to replace the invalid frame to become the voice frame to be processed and output at this time. The voice frame compensation methods applied by the device at the network side comprise one of the following: valid frame copying method, 1/4-rate frame filling method, simulation approximating method, and so on.

[0033] Valid frame copying method: the last valid frame is used to replace the current invalid frame;

1/4-rate frame filling method: this method can only be applied in EVRC encoded and decoded voice call; wherein a 1/4-rate frame with any content is used to replace the current invalid frame to be compensated;

Simulation approximating method: a frame is simulated using the rate and content of the last valid frame and the frame distance between the current invalid frame and its last valid frame according to the discipline obtained from simulation, and the frame obtained from simulation is used to replace the current invalid frame.

[0034] After compensation, the compensated voice frame is normally processed and output.

[0035] The present invention offers a system for realizing voice compensation in mobile communication network, the system is set in the device at the network side and is applied in the condition that the radio environment is bad or transmission quality is poor and the device at the network side does not use or partially use vocoder, as shown in FIG. 2, and the system comprises:

an invalid frame detection unit for judging whether the forward or backward voice frame received or to be sent by the device at the network side is an invalid frame or not at each time of processing frames; sending the invalid frame to the voice compensation unit and sending the valid frame to the unit for processing the voice frames in the device at the network side; a voice compensation unit, comprising:

a voice compensation judgment unit for receiving the invalid frame sent by the invalid frame detection unit, sending the invalid frame in the condition of non-1/8-rate whose frame distance from the last valid frame is less than or equals to compensation threshold to the voice compen-

sation process unit, and sending other invalid frame to the unit for processing voice frames in the device at the network side;

a voice compensation process unit for receiving invalid frames sent by the voice compensation judgment unit and performing voice compensation for them; that is, performing one of the following processes:

using the last valid frame to replace the current invalid frame;

using a 1/4-rate frame with any content to replace the current invalid frame; or using the frame obtained through simulation to replace the current invalid frame.

[0036] The compensated voice frames are sent to the unit for processing voice frames in the device at the network side.

[0037] Said device at the network side can be one of the following: a base station, a base station controller, a radio network controller or a mobile switching center.

[0038] The present invention can be used for voice call in which the device at the network side does not use or partially uses vocoder for voice compensation, including the mobile communication system applying TrFO technology, RTO technology or TFO (Tandem Free Operation) technology. The present invention also can be used in a mobile communication system of CDMA2000, WCDMA (Wideband-Code Division Multiple Access) and TDS-CDMA (Time Division Synchronization- Code Division Multiple Access).

[0039] In the following, the present invention will be described in further detail with three embodiments.

Embodiment 1: Using valid frame copying method to implement voice compensation.

[0040] The method for voice compensation applied in this embodiment is valid frame copying method, and the threshold of frame distance of voice compensation in this embodiment is 1, that is, voice compensation is performed only for the first invalid frame after the valid frames in the condition of full-rate, while voice compensation is not performed for the continuously arrived invalid frames after the first invalid frame; As shown in FIG. 3, the specific steps which should be performed are as follows:

101: The device at the network side judges the received forward voice frames from the network side at each time of processing the forward voice frame:

If the forward voice frame arrived at this time is an invalid frame, the process proceeds to step 102;

If the forward voice frame arrived at this time is a valid frame, the process directly proceeds to step 104;

102: The device at the network side judges the last arrived frame;

If the last frame is also an invalid frame, there is no special processing and the process directly proceeds to step 104;

If the last frame is non-full-rate frame, there is no special processing and the process directly proceeds to step 104;

If the last frame is a full-rate frame, the process proceeds to step 103; please note that the full-rate frame here is a valid frame.

103: The current invalid frame is discarded, the last arrived frame, that is, said full-rate frame, is used to replace this invalid frame; the process proceeds to step 104;

104: The current forward voice frame is normally processed and output.

[0041] Although this embodiment only describes the steps of judging and compensating the forward voice frames from the network side by the device at the network side, it is also applicable for the steps of judging and compensating the backward voice frames from the UE by the device at the network side, which will not be described in detail here.

Embodiment 2: Using 1/4-rate frame filling method to implement voice compensation.

[0042] This embodiment can be applied for EVRC encoded and decoded calls, and the legal encoding format of EVRC does not include 1/4-rate frames. A lot of experiments show that, vocoder of each UE will perform voice compensation after receiving EVRC encoded and decoded 1/4-rate frames. As shown in FIG 4, the specific steps of this embodiment are as follows:

201: The device at the network side judges the rate of the received forward voice frames from the network side at each time of processing the forward voice frame:

If the forward voice frame arrived at this time is an invalid frame, the process proceeds to step 202;

If the forward voice frame arrived at this time is a normal voice frame, the process directly proceeds to step 205;

202: The device at the network side judges the rate of the last valid frame:

If the last valid frame is a full-rate frame, the process proceeds to step 203;

If the last valid frame is non-full-rate frame, there is no special processing and the process directly

proceeds to step 205;

203: The device determines the frame distance between the last valid frame and the current invalid frame:

If the frame distance is less than or equals to the predetermined compensation threshold, the process proceeds to step 204;

Otherwise, the process proceeds to step 205;

204: The current invalid frame is discarded and a 1/4-rate frame with any content is used to replace this invalid frame. This 1/4-rate frame is used as the forward voice frame at this time. The process proceeds to step 205;

205: the current forward voice frame is normally processed and output.

[0043] From the above steps, it can be seen that the main idea of this embodiment is replacing a group of consecutive frames right after the full-rate frames with 1/4-rate frames; For each invalid frame whose frame distance from its last full-rate valid frame is less than or equals to the predetermined threshold, it is replaced with a 1/4-rate frame; No extra voice compensation is performed for those whose frame distance is greater than the threshold; that is, if the number of consecutive invalid frames right after the full-rate valid frame is greater than the maximal threshold, no extra voice compensation will be performed for those ones which exceed the threshold; said maximal threshold of the number of invalid frames is the compensation threshold. In practical applications, the compensation threshold of this method can be set as infinite, that is, all consecutive invalid frames right after the full-rate frames will be replaced with 1/4-rate frames.

[0044] Although this embodiment only describes the steps of judging and compensating the forward voice frames from the network side by the device at the network side, it is also applicable for the steps of judging and compensating the backward voice frames from UE by the device at the network side, which will not be described here.

Embodiment 3: Using simulation approximating method to implement voice compensation.

[0045] In this embodiment, a large amount of full-rate voice data are statistically inducted according to the previous actual situations to obtain the approximation rule of the content of frames and rate change; when the invalid frames are compensated, a frame can be simulated and used to replace the rate and content of the invalid frame according to the approximation rule and by using the content and rate of the last validate frame and the frame distance between the invalid frame and the last valid frame; in this specification, the frames obtained through simulation are called as pseudo-full-rate frames. The pre-

determined compensation threshold in this embodiment is 6. As shown in FIG. 5, the specific steps of this embodiment are as follows:

301: The device at the network side judges the rate of the received forward voice frames from the network side at each time of processing the forward voice frame:

If the frame arrived at this time is an invalid frame, the process proceeds to step 302;

If the frame arrived at this time is a normal voice frame, the process directly proceeds to step 305;

302: The device judges the rate of the saved last valid frame:

If the last valid frame is full-rate, the process proceeds to step 303;

If the last valid frame is non-full-rate frame, there is no special processing and the process directly proceeds to step 305;

303: The device determines the frame distance between the last valid frame and the current invalid frame:

If the frame distance is less than or equals to 6, the process proceeds to step 304;

Otherwise, the process proceeds to step 305;

304: This invalid frame is discarded, a pseudo full-rate frame is simulated according to the approximation rule obtained through statistical induction by using the content of last valid frame and the frame distance between the last valid frame and the current invalid frame as parameters, the simulated pseudo full-rate frame is used to replace this invalid frame; the pseudo full-rate frame is used as the forward voice frame at this time; the process proceeds to step 305;

305: The current forward voice frame is normally processed and output.

[0046] From the above steps, it can be seen that the main idea of this embodiment is replacing the consecutive invalid frames right after the full-rate frames with simulated voice frames, in simulation, the 6 consecutive invalid frames right after the full-rate frames can be compensated through statistical discipline according to the content of the full-rate frames and the frame distance between the invalid frame and the full-rate frame.

[0047] Although this embodiment only describes the steps of judging and compensating the forward voice frames from the network side by the device at the network side, it is also applicable for the steps of judging and compensating the backward voice frames from the UE by the device at the network side, which will not be de-

scribed here.

[0048] The above three embodiments have their own advantages, but for voice quality, the voice quality obtained through simulation approximating method is the best, and moreover, the simulation approximating method can compensate several deleted frames in the condition of consecutive full-rate frames, and its overhead is not heavy, only the content of the latest full-rate frame should be saved.

[0049] The above three embodiments mainly compensate the frames in the condition of full-rate; in practical applications, it may be supposed that voice compensation will be performed when the last valid frame is a full-rate or half-rate frame. In addition, in practical applications, the compensation threshold can be set according to the practical situations.

Industrial Applicability

[0050] The present invention offers a system and method for realizing voice compensation at the network side to solve the problem that the whole voice quality is poor and makes human ears uncomfortable because vocoder is not used or partially used in the network side for voice quality compensation and linear prediction so as to make the voice quality largely rely on whether the vocoder in the UE compensates some to-be-compensated frames and on the compensation performance. When the radio environment is bad or the network transmission quality is relatively poor, the technical scheme of the present invention can relatively compensate the voice when vocoder is not used or partially used at the network side to reduce the uncomfortable feelings of ears due to swallowing words, trilling and discontinuous voice, to promote the whole voice quality, and to reduce the dependency of calls on the performance of UE and its vocoder.

Claims

1. A method for realizing voice compensation in a mobile communication network, comprising:

- a. at each time of processing frames, a device at network side judging whether a voice frame received or to be sent is an invalid frame or not, if yes, proceeding to the next step;
- b. the device at the network side performing voice compensation for the invalid frame;

wherein there is the following step after step a:

- a1. judging if the invalid frame is a frame in the condition of non-1/8-rate or not; if yes, proceeding to the next step; and wherein a manner for judging whether the invalid frame is a frame in the condition of non-1/8-rate or not in step a1 is:

judging whether a last valid frame before the invalid frame is non-1/8-rate frame; if yes, the invalid frame is the frame in the condition of non-1/8-rate; otherwise, the invalid frame is not the frame in the condition of non-1/8-rate.

2. The method of claim 1, wherein said voice frame is a forward or backward voice frame; when the voice frame is a forward voice frame, said last valid frame is a last valid frame before the forward voice frame; when the voice frame is a backward voice frame, said last valid frame is a last valid frame before the backward voice frame.
3. A system for realizing voice compensation in a mobile communication network, wherein the system is set in a device at network side, and comprises:

an invalid frame detection unit for judging whether a voice frame received or to be sent by the device at the network side is an invalid frame or not, sending the invalid frame to a voice compensation unit and sending a valid frame to a unit for processing the voice frame in the device at the network side; and

the voice compensation unit for performing voice compensation for the invalid frame and sending the compensated voice frame to the unit for processing voice frames in the device at the network side; wherein said voice compensation unit comprises:

a voice compensation judgment unit for receiving the invalid frames sent by the invalid frame detection unit, sending invalid frames in the condition of non-1/8-rate to the voice compensation process unit and other invalid frames to the unit for processing voice frames in the device at the network side; and

a voice compensation process unit for receiving the invalid frames sent by the voice compensation judgment unit and performing voice compensation for these invalid frames; sending the compensated voice frames to the unit for processing voice frames in the device at the network side; wherein said voice compensation judgment unit is adapted to judge whether the last valid frame before the received invalid frame is a non-1/8-rate frame or not; if yes, the invalid frame is considered as an invalid frame in the condition of non-1/8-rate; otherwise, the invalid frame is not a frame in the condition of non-1/8-rate.

4. The system of claim 3, wherein said voice frame received by the device at the network side is a forward or backward voice frame;
when the voice frame is a forward voice frame, said last valid frame is a last valid frame before the forward voice frame;
when the voice frame is a backward voice frame, said last valid frame is a last valid frame before the backward voice frame.
5. The system of claim 3, wherein said device at the network side is a base station, a base station controller, a radio network controller or a mobile switching center.

Patentansprüche

1. Verfahren zum Durchführen von Sprachausgleich in einem mobilen Kommunikationsnetzwerk, Folgendes umfassend:
- a. zu jedem Zeitpunkt des Bearbeitens von Frames, Beurteilen durch eine Vorrichtung auf Netzwerkseite, ob ein empfangener oder zu sendender Sprachframe ein ungültiger Frame ist oder nicht, falls ja, Fortfahren mit dem nächsten Schritt;
- b. Ausführen von Sprachausgleich für den ungültigen Frame durch die Vorrichtung auf der Netzwerkseite; wobei nach Schritt a der folgende Schritt folgt:
- a1. Beurteilen, ob der ungültige Frame ein Frame in dem Zustand einer Nicht-1/8-Rate ist oder nicht; falls ja, Fortfahren mit dem nächsten Schritt; und wobei eine Art zum Beurteilen, ob der ungültige Frame ein Frame in dem Zustand einer Nicht-1/8-Rate ist oder nicht, in Schritt a1 Folgendes ist:
- Beurteilen, ob ein letzter gültiger Frame vor dem ungültigen Frame ein Nicht-1/8-Raten-Frame ist; falls ja, ist der ungültige Frame der Frame im Zustand einer Nicht-1/8-Rate; ansonsten ist der ungültige Frame nicht der Frame im Zustand einer Nicht-1/8-Rate.
2. Verfahren nach Anspruch 1, wobei der Sprachframe ein Vorwärts- oder Rückwärtssprachframe ist; wenn der Sprachframe ein Vorwärtssprachframe ist, ist der letzte gültige Frame ein letzter gültiger Frame vor dem Vorwärtssprachframe; wenn der Sprachframe ein Rückwärtssprachframe ist, ist der letzte gültige Frame ein letzter gültiger Frame vor dem Rückwärtssprachframe.

3. System zum Durchführen von Sprachausgleich in einem mobilen Kommunikationsnetzwerk, wobei das System in einer Vorrichtung auf Netzwerkseite eingesetzt ist und Folgendes umfasst:

eine Erkennungseinheit für ungültige Frames zum Beurteilen, ob ein durch die Vorrichtung auf der Netzwerkseite empfangener oder zu sendender Sprachframe ein ungültiger Frame ist oder nicht, Senden des ungültigen Frames an eine Sprachausgleichseinheit und Senden eines gültigen Frames an eine Einheit zum Bearbeiten des Sprachframes in der Vorrichtung auf der Netzwerkseite; und

die Sprachausgleichseinheit zum Ausführen von Sprachausgleich für den ungültigen Frame und Senden des ausgeglichenen Sprachframes an die Einheit zum Bearbeiten von Sprachframes in der Vorrichtung auf der Netzwerkseite; wobei die Sprachausgleichseinheit Folgendes umfasst:

eine Sprachausgleichsbeurteilungseinheit zum Empfangen der durch die Erkennungseinheit für ungültige Frames gesendeten ungültigen Frames, Senden ungültiger Frames im Zustand einer Nicht-1/8-Rate an die Sprachausgleichsprozesseinheit und anderer ungültiger Frames an die Einheit zum Bearbeiten von Sprachframes in der Vorrichtung auf der Netzwerkseite; und

eine Sprachausgleichsprozesseinheit zum Empfangen der durch die Sprachausgleichsbeurteilungseinheit gesendeten ungültigen Frames und Ausführen von Sprachausgleich für diese ungültigen Frames; Senden der ausgeglichenen Sprachframes an die Einheit zum Bearbeiten von Sprachframes in der Vorrichtung auf der Netzwerkseite; wobei die Sprachausgleichsbeurteilungseinheit angepasst ist, zu beurteilen, ob der letzte gültige Frame vor dem empfangenen ungültigen Frame ein Nicht-1/8-Raten-Frame ist oder nicht; falls ja, gilt der ungültige Frame als ein ungültiger Frame im Zustand einer Nicht-1/8-Rate; ansonsten ist der ungültige Frame nicht ein Frame im Zustand einer Nicht-1/8-Rate.

4. System nach Anspruch 3, wobei der durch die Vorrichtung auf der Netzwerkseite empfangene Sprachframe ein Vorwärts- oder Rückwärtssprachframe ist; wenn der Sprachframe ein Vorwärtssprachframe ist, ist der letzte gültige Frame ein letzter gültiger Frame vor dem Vorwärtssprachframe; wenn der Sprachframe ein Rückwärtssprachframe ist, ist der letzte gültige Frame ein letzter gültiger Frame vor dem Rückwärtssprachframe.

Frame vor dem Rückwärtssprachframe.

5. System nach Anspruch 3, wobei die Vorrichtung auf der Netzwerkseite eine Basisstation, eine Basisstationsteuerung, eine Funknetzwerksteuerung oder eine mobile Vermittlungsstelle ist. 5

Revendications

1. Procédé de réalisation de compensation de voix dans un réseau de communication mobile, comprenant : 10
- a. à chaque occurrence de traitement de trames, l'estimation par un dispositif côté réseau pour savoir si une trame de voix reçue ou à envoyer est une trame invalide ou non, si oui, le passage à l'étape suivante ; 15
- b. la réalisation par le dispositif côté réseau d'une compensation de voix pour la trame invalide ; dans lequel on a l'étape suivante après l'étape a : 20
- a1. l'estimation si la trame invalide est une trame dans la condition sans taux 1/8 ou non ; si oui, le passage à l'étape suivante ; et dans lequel une manière d'estimer si la trame invalide est une trame dans la condition sans taux 1/8 ou non à l'étape a1 est : 25
- l'estimation permettant de savoir si une dernière trame valide avant la trame invalide est une trame sans taux 1/8 ; si oui, la trame invalide est la trame dans l'état sans taux 1/8 ; sinon, la trame invalide n'est pas la trame dans la condition sans taux 1/8. 30
2. Procédé selon la revendication 1, dans lequel ladite trame de voix est une trame de voix vers l'avant ou vers l'arrière ; 40
- lorsque la trame de voix est une trame de voix vers l'avant, ladite dernière trame valide est une dernière trame valide avant la trame de voix vers l'avant ; 45
- lorsque la trame de voix est une trame de voix vers l'arrière, ladite dernière trame valide est une dernière trame valide avant la trame de voix vers l'arrière.
3. Système de réalisation de compensation de voix dans un réseau de communication mobile, dans lequel le système est installé dans un dispositif côté réseau, et comprend : 50
- une unité de détection de trame invalide pour estimer si une trame de voix reçue ou à envoyer par le dispositif côté réseau est une trame invalide ou non, envoyer la trame invalide à une unité 55

de compensation de voix et envoyer une trame valide à une unité pour traiter la trame de voix dans le dispositif côté réseau ; et l'unité de compensation de voix pour réaliser une compensation de voix pour la trame invalide et envoyer la trame de voix compensée à l'unité pour traiter les trames de voix dans le dispositif côté réseau ; dans lequel ladite unité de compensation de voix comprend :

une unité d'estimation de compensation de voix pour recevoir les trames invalides envoyées par l'unité de détection de trame invalide, envoyer des trames invalides dans la condition sans taux 1/8 à l'unité de traitement de compensation de voix et d'autres trames invalides à l'unité pour traiter des trames de voix dans le dispositif côté réseau ; et une unité de traitement de compensation de voix pour recevoir les trames invalides envoyées par l'unité d'estimation de compensation de voix et réaliser une compensation de voix pour ces trames invalides ; envoyer les trames de voix compensées à l'unité pour traiter des trames de voix dans le dispositif côté réseau ; dans lequel ladite unité d'estimation de compensation de voix est adaptée pour estimer si la dernière trame valide avant la trame invalide reçue est une trame sans taux 1/8 ou non ; si oui, la trame invalide est considérée comme une trame invalide dans l'état sans taux 1/8 ; sinon, la trame invalide n'est pas une trame dans l'état sans taux 1/8.

4. Système selon la revendication 3, dans lequel ladite trame de voix reçue par le dispositif côté réseau est une trame de voix vers l'avant ou vers l'arrière ; lorsque la trame de voix est une trame de voix vers l'avant, ladite dernière trame valide est une dernière trame valide avant la trame de voix vers l'avant ; lorsque la trame de voix est une trame de voix vers l'arrière, ladite dernière trame valide est une dernière trame valide avant la trame de voix vers l'arrière.
5. Système selon la revendication 3, dans lequel ledit dispositif côté réseau est une station de base, un dispositif de commande de station de base, un dispositif de commande de réseau de radiocommunication ou un centre de commutation mobile.

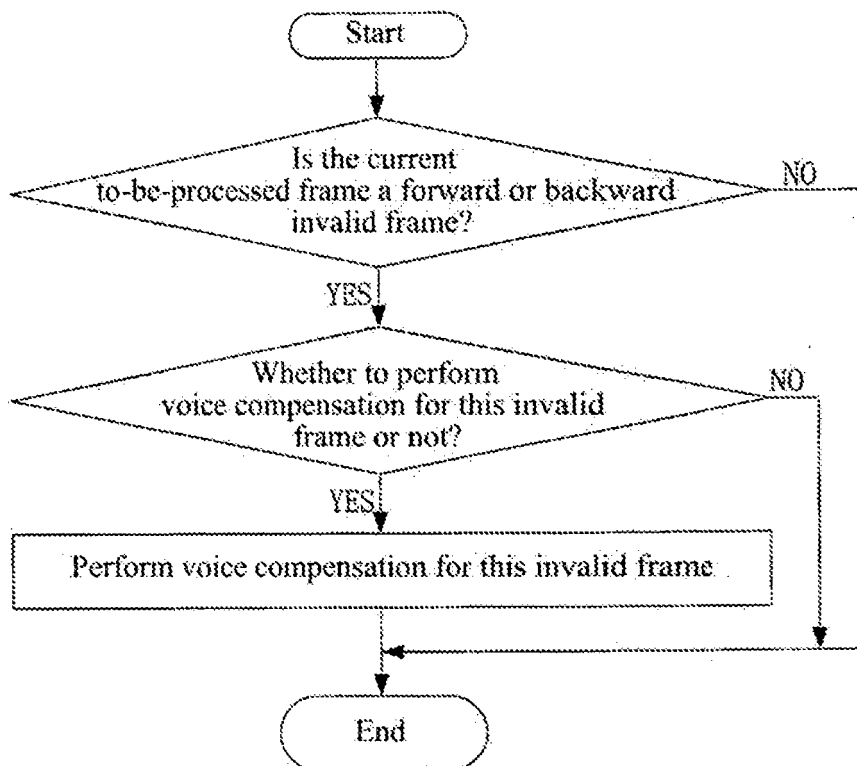


FIG. 1

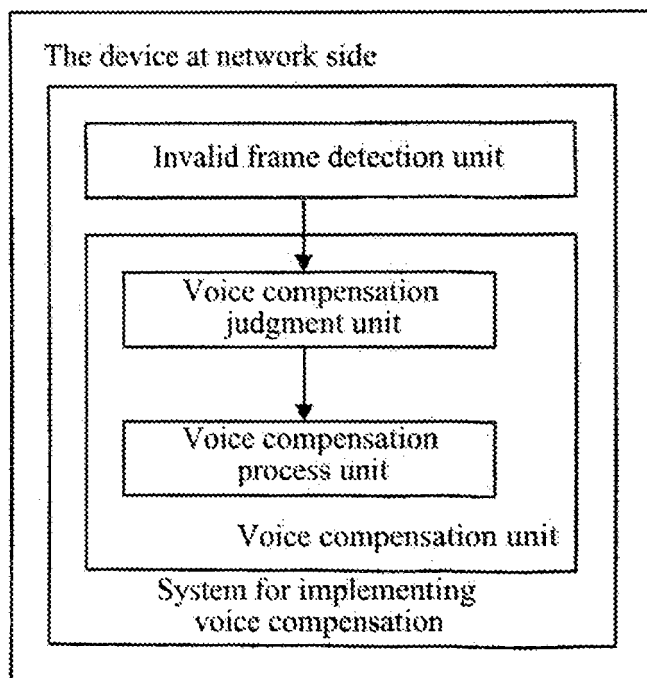


FIG. 2

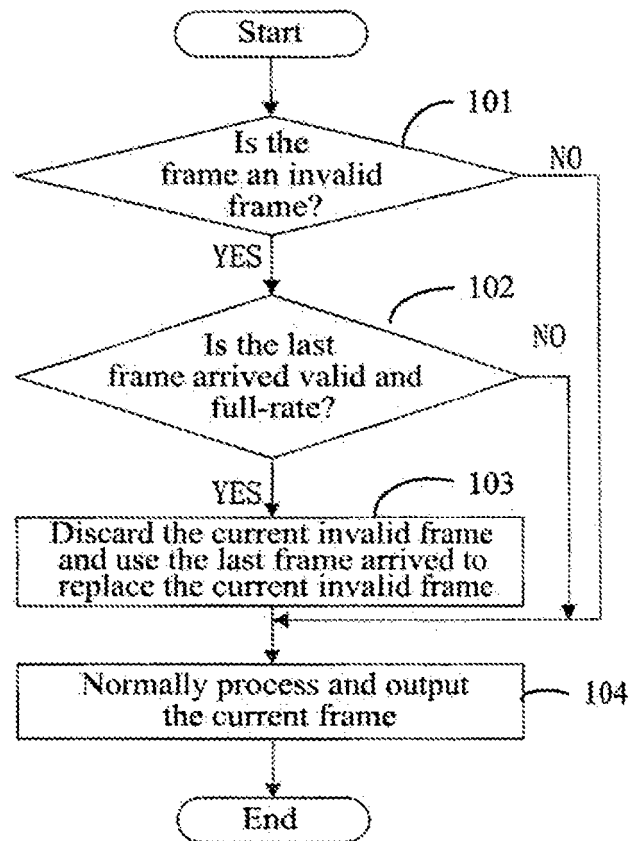


FIG. 3

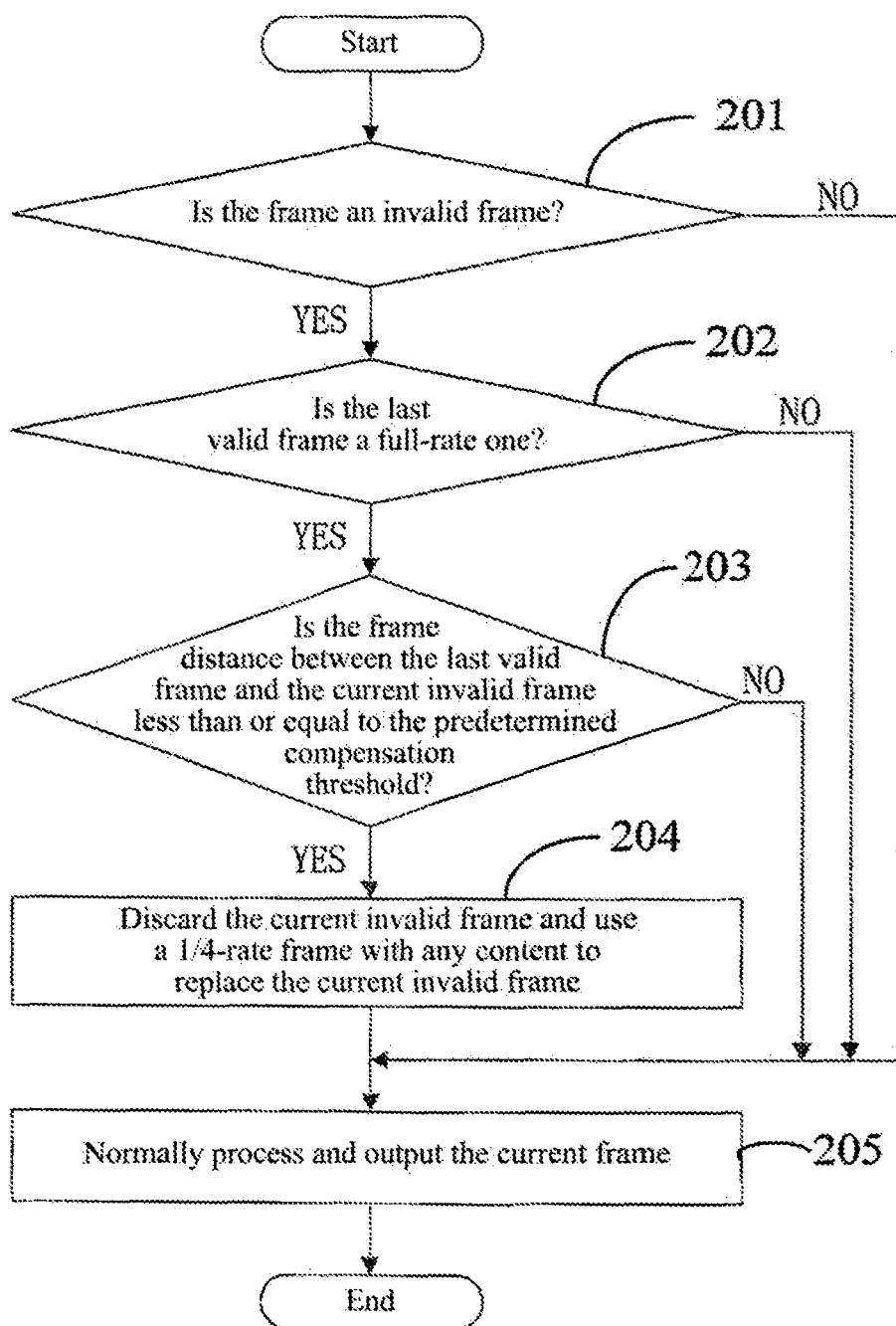


FIG. 4

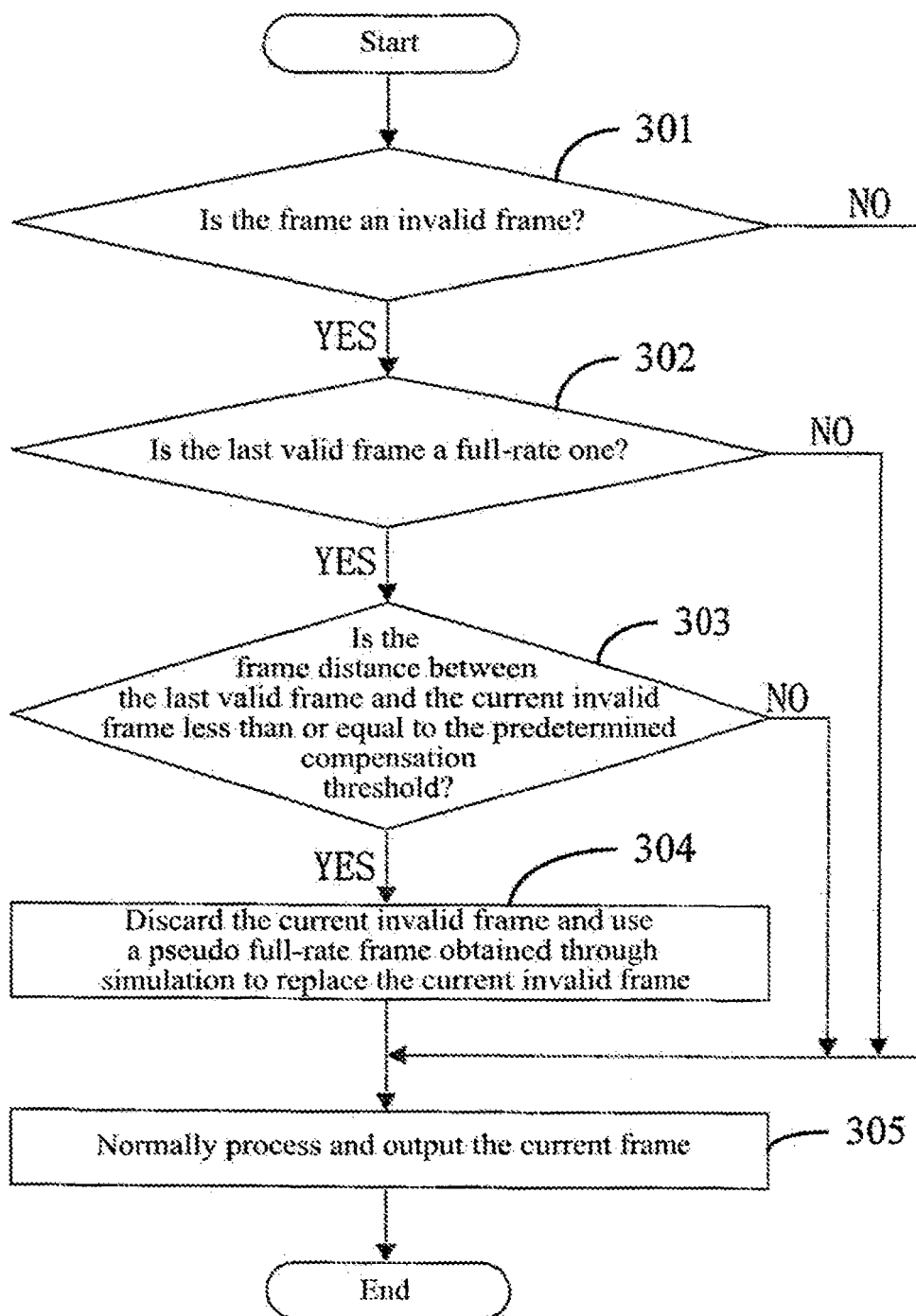


FIG. 5

REFERENCES CITED IN THE DESCRIPTION

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