Claims:

1. A method for jitter compensation of random temporal fluctuations, which are termed jitter (JIT), between the times of receipt (te) of voice packets (110, 120; 210) when receiving voice content (1, 2, 3, 4) via IP-based networks, in which a sequence of digital voice packets (110, 120; 210) with or without voice content (1, 2, 3, 4) is received by a receiver for processing voice content (1, 2, 3 4) contained in the voice packets (110, 120; 210),

a voice packet (110; 210) with voice content having a section, termed the header (H), with data for communication control and a section, termed the payload (PL), with digital voice data from part of the voice content (1, 2, 3, 4),

a voice packet (120) without voice content having the header (H) without the payload,

at least one piece of time information of a transmitter time of a transmitter being contained in the voice packet (110, 120; 210), wherein the piece of time information indicates the time of sending (ts) of the voice packet (110, 120; 210), wherein the sampling instant of a first sample when sampling the voice content (1, 2, 3, 4) is adopted as the time of sending of the voice packet (110, 120, 210), and

the receiver waiting to process the voice content (1, 2, 3, 4) for a wait time, termed a buffer (DJB), after receiving a first voice packet (111) with voice content (1, 2, 3, 4),

characterized

in that, after the receipt of a first voice packet (110, 120; 210), termed the initial packet, during a connection period (30) on a transmission channel (10, 20) of the voice packet (110, 120; 210), the receiver initializes a receiver clock using a transmitter time by means of the time information in the initial packet, so that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210),

in that, after the receiver clock has been initialized, the receiver determines a minimum relative packet transmission duration (*delta*) for each receipt of subsequent voice packets (110, 120; 210) by

- Comparing (304) the receiver time with the time information at the transmitter time when the voice packet was sent out, which is contained in the header (H) of the subsequent voice packet (110, 120; 210), and determining (305) a relative packet transmission duration (*delta*) relative to the receiver time,
- temporarily storing (306) a resultant minimum relative packet transmission duration (*delta*) , and
- adjusting (308) the receiver time dependent on the temporarily stored minimum relative packet transmission duration (*delta*) in such a manner that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210) with the minimum relative packet transmission duration,

in that, when receiving a first voice packet (111) with voice content after at least one voice packet (120) without voice content, the receiver determines the actual time of receipt of this voice packet by comparing (*deltaJB*) the time information in the received voice packet (111) with the receiver time at receipt and determines the buffer (DJB) as a function of the actual time of receipt.

2. The method according to claim 1, **characterized** in that during the initialization of the receiver clock and during the determination of the relative packet transmission duration (*delta*), a different length of voice packets (110; 210) with voice content and voice packets (120) without voice content is taken into account.

3. The method according to claim 1 or 2, **characterized** in that the determination of the minimum packet transmission duration (*delta*) takes place at multiple intervals during the connection period (30).

4. The method according to one of the preceding claims, **characterized** in that the adjustment of the receiver time as a function of the temporarily stored minimum relative packet transmission duration (*delta*) takes place in such a manner that the receiver time is advanced by the minimum relative packet transmission duration (*delta*), if this indicates that a voice packet (110, 120; 210) with a shorter packet transmission time was received.

5. The method according to the preceding claims, **characterized** in that the adjustment of the receiver time as a function of the temporarily stored minimum relative packet transmission duration (*delta*) takes place in such a manner that the receiver time is set back if the temporarily stored minimum relative packet transmission duration (*delta*) indicates that only voice packets with a longer packet transmission duration were received, wherein the receiver time is set back by a defined or definable duration.

6. The method according to one of the preceding claims, **characterized** in that the determination of the minimum relative packet transmission duration (*delta*) takes place for voice packets (120) without voice content and voice packets (110; 210) with voice content.

7. The method according to one of the preceding claims, **characterized** in that the buffer (DJB), starting from a maximum buffer, adjusts a maximum jitter to be taken into account on the basis of a delay to receipt (*deltaJB*) resulting from the comparison of the time information in the first voice packet (111) with voice content and the receiver time when this first voice packet (111) is received.

8. The method according to one of the preceding claims, **characterized** in that the voice packet (110; 210) with identical voice content is received by the receiver via a plurality of different transmission channels (10, 20), wherein a correction of the delay (DDC) of the voice packets (110, 210) with identical voice content via the various transmission channels is applied by the receiver.

9. A receiver for receiving digital voice packets (110, 120; 210) via IP-based networks and for processing voice content (1, 2, 3, 4) contained in the voice packets (110, 120; 210),

said voice packets (110, 120; 210) being voice packets (110; 210) with voice content having a section, termed the header (H), with data for communication control and a section, termed the payload (PL), with digital voice data from part of the voice content (1, 2, 3, 4) or voice packets (120) without voice content having the header (H) without the payload, wherein at least one piece of time information of a transmitter time of a transmitter is contained in the voice packet (110, 120; 210), which piece of time information indicates the time of sending (ts) of the voice packet (110, 120; 210), wherein the sampling instant of a first sample when sampling the voice content (1, 2, 3, 4) is adopted as the time of sending of the voice packet (110, 120, 210),

having a receiving unit, which can be connected to the IP-based network and is set up for receiving voice packets (110, 120; 210) transmitted via the IP-based network, and

having an arithmetic logic unit with receiver clock, which is set up to process the received voice packets (110, 120; 210),

wherein the arithmetic logic unit is set up to execute a method for jitter compensation when receiving and processing the voice content (1, 2, 3, 4) as follows:

waiting to process the voice content (1, 2, 3, 4) for a wait time, termed a buffer (DJB), after receiving a first voice packet (111) with voice content (1, 2, 3, 4),

after the receipt of a first voice packet (110, 120; 210), termed the initial packet, during a connection period (30) on a transmission channel (10, 20) of the voice packet (110, 120; 210), initializing a receiver clock using a transmitter time by means of the time information in the initial packet, so that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210),

after the receiver clock has been initialized, determining a minimum relative packet transmission duration (delta) for each receipt of subsequent voice packets (110, 120; 210) by

- comparing (304) the receiver time with the time information at the transmitter time when the voice packet was sent out, which is contained in the header (H) of the subsequent voice packet (110, 120; 210), and determining (305) a relative packet transmission duration (delta) relative to the receiver time,
- temporarily storing (306) a resultant minimum relative packet transmission duration (delta), and

- adjusting (308) the receiver time dependent on the temporarily stored minimum relative packet transmission duration (delta) in such a manner that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210) with the minimum relative packet transmission duration,

when receiving a first voice packet (111) with voice content after at least one voice packet (120) without voice content, determining the actual time of receipt of this voice packet by comparing (deltaJB) the time information in the received voice packet (111) with the receiver time at receipt and determining the buffer (DJB) as a function of the actual time of receipt.

10. The receiver according to claim 9, **characterized** in that the receiver is set up to receive the voice packet (110; 210) with the identical voice content via a plurality of different transmission channels (10, 20) and to apply the method according to Claim 8 in the process, wherein the arithmetic logic unit is set up to process the voice content (1, 2, 3, 4) from the received voice packets (110, 120; 210) with the identical voice content from a plurality of the various transmission channels (10, 20).

11. A method for sending and receiving voice content,

in which a transmitter converts voice content (1, 2, 3, 4) into a sequence of digital voice packets (110, 120; 210) with or without voice content, a voice packet (110; 210) with voice content having a section, termed the header (H), with data for communication control and a section, termed the payload (PL), with voice data digitized from part of the voice content (1, 2, 3, 4) and a voice packet (120) without voice content having the header (H) without the payload, and each voice packet (110, 120; 210), particularly the header (H) of the voice packet (110, 120; 210), containing at least one piece of time information of a transmitter time, which

indicates the time of sending (ts) of the voice packet (110, 120; 210)), wherein the sampling instant of a first sample when sampling the voice content (1, 2, 3, 4) is adopted as the time of sending of the voice packet (110, 120, 210), and sends out the sequence of the voice packets (110, 120; 210) via at least one transmission channel (10, 20) of an IP-based network, and

in which a receiver receives the sequence of digital voice packets (110, 120; 210) with or without voice content and processes the voice content (1, 2, 3, 4),

said receiver carrying out the steps of

waiting to process the voice content (1, 2, 3, 4) for a wait time, termed a buffer (DJB), after receiving a first voice packet (111) with voice content (1, 2, 3, 4),

after the receipt of a first voice packet (110, 120; 210), termed the initial packet, during a connection period (30) on a transmission channel (10, 20) of the voice packet (110, 120; 210), initializing a receiver clock using a transmitter time by means of the time information in the initial packet, so that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210),

wherein the method is characterized in that after the receiver clock has been initialized, determining a minimum relative packet transmission duration (delta) for each receipt of subsequent voice packets (110, 120; 210) by

- comparing (304) the receiver time with the time information at the transmitter time when the voice packet was sent out, which is contained in the header (H) of the subsequent voice packet (110, 120; 210), and

determining (305) a relative packet transmission duration (delta) relative to the receiver time,

- temporarily storing (306) a resultant minimum relative packet transmission duration (delta), and
- adjusting (308) the receiver time dependent on the temporarily stored minimum relative packet transmission duration (delta) in such a manner that the receiver time correlates with the transmitter time when sending out the voice packet (110, 120; 210) with the minimum relative packet transmission duration,

when receiving a first voice packet (111) with voice content after at least voice packet (120) without voice content, determining the actual time of receipt of this voice packet by comparing (deltaJB) the time information in the received voice packet (111) with the receiver time at receipt and determining the buffer (DJB) as a function of the actual time of receipt.

12. The method according to claim 11, **characterized** in that a permanent communication connection is set up in the IP-based network for a connection period (30) between the transmitter and the receiver via at least one communication channel (10, 20), wherein, during the connection period (30), at times when no voice content (1, 2, 3, 4) should be transmitted, voice packets (120) without voice content are exchanged to maintain the connection between the transmitter and receiver, and wherein, at times during the connection period (30) when voice content (1, 2, 3, 4) should be transmitted, the sequence of digital voice packets (110; 210) with the voice content is transmitted from the transmitter to the receiver.

13. The method according to claim 11 or 12, **characterized** in that the transmitter sends out a voice packet (110, 120; 210) via a plurality of different transmission channels (10, 20) and the receiver receives the voice packets (110, 120; 210) sent out via various transmission channels (10, 20).

14. The method according to one of claims 11 to 13, **characterized** in that the method is used in radio communication, particularly for the transmission of operational ground-to-air communication in CLIMAX operation, in which a plurality of different transmission channels (10, 20) are used for transmitting a radio message as voice content (1, 2, 3, 4).

15. A device for sending and receiving digital voice packets (110, 120; 210) via IP-based networks with voice content (1, 2, 3, 4) contained in the voice packets (110; 210) having a transmitter and a receiver,

the transmitter having a voice recording unit for recording the voice content (1, 2, 3 4), and an arithmetic logic unit with transmitter clock, the arithmetic logic unit being set up for processing the voice content (1, 2, 3, 4) according to the method according to one of claims 11 to 14, and

the receiver being constructed according to the features according to one of claims 9 or 10.