(54) Titre : PROCEDE DE TRAITEMENT D'UN SIGNAL STEREO FM
(54) Title: METHOD FOR PROCESSING AN FM STEREO SIGNAL
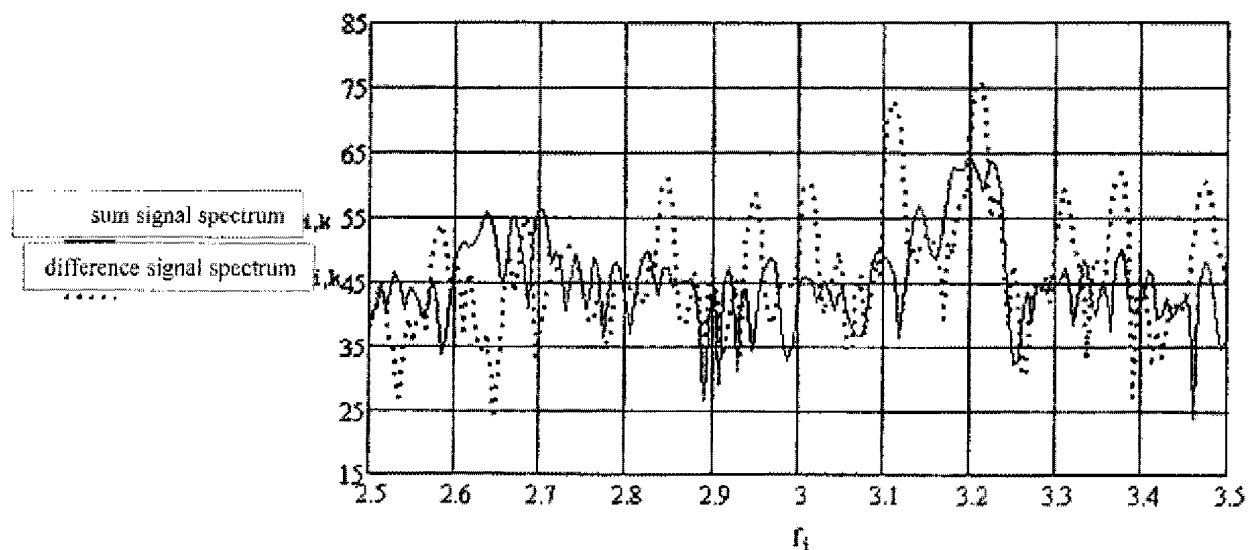
(57) Abrégé/Abstract:

A method for processing an FM stereo signal. The FM stereo signal is digitized and divided into overlapping blocks, which are transformed into the frequency domain. Individual spectral lines of the difference signal are lowered if these have a higher magnitude than the respective spectral lines of the sum signal. The sum and difference signals are then transformed back.

Method for processing an FM Stereo Signal

5   Field of the invention

The invention relates to a method for processing an analog FM
stereo signal subjected to digital signal processing.

10   Background

The pilot-tone system described in ITU-R BS.450 is used to
transmit stereo signals from the FM transmitter. This system
applies a preemphasis (high frequency boost) to the left (L)
15   and right (R) audio channels, before using a matrix which
generates a sum signal (L+R)/2 and a difference signal (L-R)/2.

The sum signal is transmitted in base band up to 15 kHz. The
difference signal is transmitted in double sideband modulation,
20   with the 38 kHz carrier suppressed. To enable the receiver to
demodulate the difference signal, a pilot tone is transmitted
with a frequency of 19 kHz, which corresponds to half the
carrier frequency.
The signal mixture of sum-, difference, and pilot-tone signal
25   is referred to as a multiplex signal (MPX). The MPX signal and
additional signals, if necessary (such as RDS) modulate an FM
transmitter's high-frequency carrier signal in its frequency.
The high-frequency broadcasting is done via an antenna.

A superheterodyne FM receiver receives the high-frequency signal via an antenna. The radio-frequency signal (RF signal) of the antenna is amplified, preselected in the frequency, and moved into an intermediate frequency (IF)

5   range.

An intermediate frequency filter lets through most of the usable signal bandwidth and filters out most adjacent channel interference. By subsequent amplitude limiting of the intermediate frequency signal in a limiter, the

10   amplitude fluctuations in the RF- and/or IF-signal reception are suppressed.

Subsequently, a frequency demodulation takes place that delivers the MPX signal. This is fed into a stereo decoder.

15

A block diagram of an MPX stereo decoder is shown in Fig. 1.

A mono receiver evaluates only the sum signal (L+R)/2 in

20   the baseband extending up to 15 kHz. In a stereo receiver, a stereo decoder obtains the L and R signals from the MPX signal.

25   In the stereo decoder, a frequency doubling of the pilot tone signal takes place, and hence a recovery of the carrier frequency 38 kHz of the difference signal occurs. The stereo decoder demodulates the double sideband-modulated difference signal and thus recovers the signal

30   (L-R)/2. The sum signal (L+R)/2 is recovered directly from the baseband. By dematrixing, meaning addition or subtraction of these two signals, the decoder recovers the

preemphasized L and R signals again. These are then
subjected to a deemphasis that compensates for the
transmitter-side preemphasis. The original signals L and R
are thus available.

5

Other decoding methods, such as the switching-decoder,
differ from the above-depicted signal processing with
regard to demodulation and dematrixing; however, they can
be converted in the above model as seen in signal theory.

10

The receiver behavior according to current technology is as
follows.

15   The FM pilot tone system should first be considered in
theory with respect to noise.

The constant noise density in the RF- or IF-range is
converted by the FM-demodulation process into a frequency-
20   proportional voltage-density.

The MPX-spectrum and noise voltage density
(Rauschspannungsdichte) are shown in Fig. 2.

25   It can be seen from Fig. 2 that the difference signal
contains significantly more noise between 23 and 53 kHz
than the sum signal, which only reaches up to 15 kHz.

The monaural audio signal-to-noise ratio SNR_{FM}, prevailing
30   after the FM demodulation with respect to +/- 75 kHz
frequency deviation without consideration of a

pre/deemphasis, can be approximated by the following
formula:

$SNR_{FM} = 3\beta^2(\beta+1)\ CNR$ with the radio-frequent carrier-to-

5 noise-ratio

$CNR = A^2/(2B_TN_0)$

β is the FM modulation index

A is the amplitude of the carrier signal

$N_0/2$ is the two-sided spectral noise power density with

10 white noise

B_T is the radio frequency transmission bandwidth

It can be estimated using the Carson formula by way of

$B_T = 2(\beta+1)W$

W is the audio signal bandwidth

15

The result obtained with the Carson formula is $\beta+1 = B_T/2W$

Used in the formula for SNR_{FM} results for $\beta \gg 1$ in

$SNR_{FM} = 3CNR(B_T/2W)^3$

20

The above-mentioned formulas apply above the so-called FM
threshold, below which the signal quality decreases rapidly
and impulse-noise can be expected, which results in clicks
or crackling after demodulation.

25

The FM threshold at a radio frequency transmission
bandwidth of 180 kHz is approximately 11 dB CNR. Above this
threshold is:

30    $SNR_{FM} = 28\ dB + CNR$ with mono reception

$SNR_{FM} = 5\ dB + CNR$ with stereo reception

10 dB can further be added when considering a preemphasis /
deemphasis of 50 μs respectively 13 dB at 75 μs.

5   The FM threshold of approximately 11 dB corresponds to a
mono audio-to-noise ratio of 39 dB + deemphasis-gain. In
case of a deemphasis of 50 μs there is likely to be at
least a 49 dB mono audio signal-to-noise ratio, or 26 dB
stereo audio signal-to-noise ratio. With regard to a 40 kHz

10   frequency deviation, an audio signal-to-noise ratio of 43.5
dB mono and 20.5 dB stereo is to be expected. The mono-gain
in the audio signal-to-noise ratio at the FM threshold is
23 dB. In the receiver, the mono-gain [N(mono) - N(stereo)]
decreases with increasing audio signal-to-noise ratio, as

15   can be seen from the limiter curve of an exemplary FM
receiver shown in Fig. 3.

The audio signal-to-noise ratio SNR is limited upwards by
the inherent noise of the rest of the transmission chain.

20   In FIG 3 the solid curve N (stereo) shows the size of the
noise N of an FM stereo reception. The dashed line N shows
the function "stereo blend" which reduces the level of the
difference signal
according to a falling of the antenna input voltage below a

25   threshold (here about 100 μV antenna voltage). The noise
power N is kept at a reduced level and does not rise
further. The result is an increasing deterioration of the
L-R channel separation (stereo blend) up to mono (L=R,
i.e., no channel separation).

From approximately 40 µV, the useful signal reaches its full level. The distance from the curve N to the curve S+N is the audio signal-to-noise ratio.

5   According to current technology, a reduction of the level of the difference signal is used to raise the audio signal-to-noise ratio at the expense of the L-R channel separation. The reduction can be made broadband or in frequency ranges, such as in the high frequencies, and

10   depends on the extent of external signals, external criteria, or an estimate of the interference signal.

Further actions in the receiver to reduce the audibility of interferences in the audio frequency range or MPX range are

15   lowering of the higher audio frequencies (hi-blend, hi-cut) during strong noise, and volume-reduction or muting (muting, noise blanker) during strong interference. These also have an effect on the sum signal (mono signal).

20

Object of the invention

Given this background, it is an object of the invention to improve the audible stereo audio signal-to-noise ratio

25   without limiting the L-R-channel separation further. The improved signal-to-noise ratio should be, in particular, the mono-quality.

Furthermore, the invention relates to the steps of

30   improvement from the signal itself without the help of

external signals or external criteria (such as the antenna
voltage).

5   Summary of the invention

The object of the invention is already achieved by a method
for processing an FM stereo signal according to one of the
independent claims.

10

Preferred embodiments and further refinements of the
invention are subject matter of the dependend claims, the
descriptions as well as the drawings.

15   The invention relates to a method for processing an analog
FM stereo signal. The invention therefore relates to the
processing of analog radio signals broadcasted in stereo
sound.

20   The analog stereo signal can be used in the manner
described in the introduction as a sum and difference
signal as well as a signal in which the left and right
channels are separated from each other.

25   The method according to the invention for processing an FM
stereo signal can also be characterized in that an FM
stereo signal is digitized, the digitized signal is
subjected to a signal processing, and is then transformed
back into the analog signal.

30

According to the invention, the signal processing is performed digitally, i.e. the processing of the signal is performed by means of a microprocessor in digital form.

5   For this purpose, the analog FM stereo signal, as sum and difference signal, is digitized. This digitized signal is divided into overlapping blocks.

The block length is preferably greater than the time shift

10   between the sum and the difference signal.
In particular, the block length can be between 10 and 1000 ms, preferably between 50 and 150 ms. A large block length leads to a high frequency resolution, but allows for a rather reduced noise reduction because the useful signal

15   spectrum approximates the interference signal spectrum.

The overlapping blocks allow a consecutive transformation into the frequency domain, and there, a signal processing.

20   After conversion, the signal is available as a difference and sum channel spectrum, in which the signal, blockwise, is divided into a plurality of spectral lines representing the magnitude and the phase of the signal at the respective frequency.

25

Thus, a first aspect of the invention relates to a digital signal processing, in which the analog signal is digitized and at least partially processed in the frequency domain. For this purpose, the

30   spectral lines of the differential signal spectrum are compared with the corresponding spectral lines of the sum signal spectrum. It is understood that under "signal

spectrum" in each case the signal magnitude spectrum is understood, and thus the comparison refers to the respective amounts. For further explanations, these are considered to be

5   logarithmic.

According to the invention spectral lines of the difference signal, spectrum are attenuated if they have a higher amount than the respective spectral lines of the

10   sum signal spectrum.

Preferably, the lowering is made to the amount of the sum signal spectrum. However, it can remain a difference to the amount of the sum signal spectrum, in particular a

15   difference of a maximum of +/- 6 dB, preferably +/- 3 dB.

This aspect of the invention is based on the assumption that the stereo signal comprises only signal components that can be localized within the stereobase.

20   It follows that a spectral line of the difference signal spectrum cannot have a higher magnitude than the corresponding spectral line of the sum signal spectrum. The conclusion applies to intensity stereophony (simultaneous sum and difference signal). It applies to time-based

25   stereophony if the time difference between the two signals or their portions do not lead to altered magnitude spectra. The latter is true if the block length clearly exceeds the time difference.

30   By transforming into the frequency domain, a processing of the signal components in digitized form is possible in a simple way. In particular, this can be done purely via

software ,for example, on a smartphone or consumer electronics device. Preferably, the entire signal processing is done purely via software, also including the processing in the time domain.

After this digital signal processing, the sum and difference signal is transformed back and the overlapping blocks are added.

The signal can now be dematrixed and converted into an analog signal for the drive of a loudspeaker.

According to the further rules, that are also described in concrete terms, in particular, interference-induced drops of the sum signal as well as interference - related increases of the spectra can be recognized and exceptions can be defined not to falsify the (undisturbed) signal and to reduce interference if necessary.

In a further embodiment of the invention, the frequency corresponding spectral lines of the difference signal spectrum are not attenuated at all or are less attenuated, if the width of a relative minimum of the sum signal spectrum is below a threshold and the depth is above a threshold, thus creating a cancellation.

This approach to the reduction of the difference signal is based on the consideration of time differences of the signals, which lead to interferences.

Interference leads to level drops (cancellations) and / or level increases at specific points of the

frequency spectrum.

However, such drops have a narrow bandwidth. If the
interference-related drops were used to cause an attenuation of
the difference signal at these points, the FM stereo signal
would be distorted.

In a further embodiment of the invention, in the range of a
local maximum of the difference signal spectrum, the latter is
compared with the sum signal spectrum, and the difference
signal spectrum is not attenuated in this range or is less
attenuated, if a maximum of the sum signal spectrum lies within
the frequency bandwidth of the maximum of the difference signal
spectrum.

This further embodiment of the invention relates to the
treatment of local maxima, which can, as well as local minima,
be caused by time delay phenomena.

Cancellations and local maxima of a frequency spectrum can be
identified as such by their frequency bandwidth and their
distance to the spectral substitute value.
A median filtering of the spectrum produces a reference curve.
If the distance of the spectrum to the reference curve exceeds
a threshold value, a cancellation or a local maximum is
identified. Their frequency bandwidth corresponds to the number
of consecutive spectral lines which exceed the threshold value.

The median value is calculated from the spectral values of the spectrum within a window around a frequency. It serves here as a substitute value in the spectrum and bridges

5   outliers of the spectrum level.

For cancellations, local maxima of the sum spectrumand local maxima of the difference signal spectrum, distinct threshold values are provided in dB, which a distance must

10   exceed in order to confirm an identification.

If, therefore, such a cancellation is confirmed, the lowering of the corresponding spectral lines of the difference signal spectrum is not made according to the

15   previously defined rule. That is, an exception is made to the previously defined rule.

A further embodiment of the invention provides that time differences between the sum and difference signals are

20   determined via a similarity analysis or a correlation of the signals in the time domain.

Specifically, by cross-correlation and/or consideration of the cancellation in the frequency domain, in particular by

25   comparing the respective frequencies and the respective bandwidths of the cancellations in the sum and difference signal spectra, the presence of time-based stereophonic portions in the signals can be determined. This is true if cancellations occur in the sum and difference signals at

30   different frequencies.

In a further embodiment of the invention, in a block-wise similarity analysis, the analysis result of the previous block is taken over if the currently processed block shows a signal-to-noise ratio (SNR) lying below a threshold value. For

5   example, if the signal-to-noise ratio falls below a threshold value in a block, the IS/LS analysis becomes unreliable. Therefore, according to this embodiment, the decision of the last block with a high signal-to-noise ratio is adopted.

10   If the calculation yields the presence of time-based stereo signal portions, an exception rule is defined. This ensures that the exeption rule is only used with time-based stereophony.

15   The further developments of the invention serve to avoid sound distortions, in particular to prevent a fault-free signal from being corrupted in an audible manner.

In a further embodiment of the invention, in the case of a
20   cancellation of a spectral line in the sum signal spectrum, the spectral line of the difference signal is only reduced to a spectral substitute value, in particular to a median value of the sum signal spectrum.

25   This is a differentiation of the cancellation rule.

If the magnitude of the difference signal spectrum at a frequency is greater than the corresponding value of the sum signal spectrum, the difference signal spectrum is

lowered in case of a cancellation to the median of the sum signal spectrum determined for this frequency.

The reduction to the spectral substitute value of the sum signal spectrum reduces interference-induced increases in

5   the difference signal spectrum and avoids a sound distortion by an otherwise greater reduction of the spectral line.

In a further embodiment of the invention, the difference

10   signal in the time domain is limited to the envelope curve of the sum signal whose maxima are held (peak hold) for a period of time before and after the entry time of the maximum. The so-changed envelope curve or the sum signal is muliplied by a factor of greater than 1, in particular a

15   factor between 1 and 2.

This aspect of the invention is based on the assumption that the stereo signal does not map spots outside the stereo base. It follows that the difference signal cannot

20   have a higher magnitude than the sum signal multiplied by a factor which takes into account a statistical signal increase which can occur with certain signal constellations of the sum and difference signals.

25   Preferably, the restriction of the difference signal in the time domain to the envelope curve of the sum signal is only performed when a transient character of the signal is detected via an evaluation of the signal or the envelope curve of the sum signal.

30

This is preferably done before the attenuation of individual spectral lines of the difference signal, that is to say, before processing in the frequency domain.

5   The envelope curve can be multiplied by a factor, in particular an empirical factor between 1.1 and 2.0, preferably between 1.3 and 1.6.

The invention further relates to a computer program which
10 includes a plurality of instructions which can be stored on a computer, in particular on a smartphone or consumer electronic device. The instructions, when processed by a microprocessor or microcontroller, perform a method as described above.
15

The invention relates in particular to a purely software-based processing of an analog FM stereo signal in a device such as, for example, a smartphone, but also in a radio with digital signal processing, in particular a car radio.
20

It is clear that, in the case of known devices in which digital signal processing is already present, all necessary further process steps, if appropriate, can be integrated into this digital processing.
25

In particular, in the case of smartphones and consumer electronics devices which comprise a microprocessor, the hardware components required for carrying out the method according to the invention are present when the device is
30 provided with a tuner for receiving analog FM stereo signals.

The method according to the invention can be implemented, in particular, purely through software, via a program (app). The instructions for carrying out the method according to the invention are stored on a data store.

5

In a further embodiment of the invention, the signal processing takes place in an application-specific integrated circuit (ASIC), which carries out the signal processing according to the method according to the

10   invention.

Description of an exemplary embodiment

15   The method according to the invention will be explained in detail below with reference to an exemplary embodiment and with reference to the further drawings.

20   1. Overview of the method

In which form (time domain or frequency domain) and to which extent the difference signal may differ from the sum signal without restricting the stereo base, is

25   theoretically derived in the following manner. The resulting rules for signal processing allow the disturbed difference signal to be approached audibly to the undisturbed difference signal without explicit knowledge of the interfering signal or without external information. The

30   interfering signal does not need to be estimated.

The rules are obtained exclusively from the signals L and R or (L + R), and (L - R). Therefore, it is also possible to process recorded stereo signals of an FM stereo receiver after the fact.

5

The signal processing of the difference signal according to the derived rules will result in an approximation of the undisturbed difference signal.

10 Thereby the achieved noise reduction is not dependent on a signal threshold. It works in all signal level ranges.

The method yields the effect of individual frequency portions within the critical bandwidth to the hearing as
15 well as the masking effect.

The modular structure of this method allows for different quality levels dependent on the varying realization efforts of signal processing to be implemented.

20

The method according to the invention for the signal processing of an FM stereo signal processes audio signals of the left and right channel after a stereo decoding in the receiver in digitized form. The sum signal and
25 difference signal can alternatively be processed.

It is assumed that functions such as stereo blend and hi-blend are turned off, and the muting level and the volume reduction in the case of severe disturbances is adapted to
30 the method, to fully exploit the benefits of the method.

The method adapts the signal processing to the signal characteristics.

To this end, various signal analyses are carried out. The aim will be for the undisturbed signals to remain practically audibly unchanged, while the disturbed signals will be freed very effectively of interference while preserving the L-R channel separation.

2. Block Structure and Overlap-add

Signal processing takes place in blocks, which means the audio data of both channels is collected for a period of time and then processed. Signal processing is non-linear and takes place in the time and frequency domain. In the following, the term "frequency domain" stands for the domain of the transformed signal. The transformation may be a Fourier transform or a wavelet transformation or the like.

The signal processing steps for noise reduction are embedded in a weighted-overlap-add structure (WOLA). It's possible to perform a consecutive transformation into the frequency domain by WOLA. The WOLA structure used consists of the following parts:

• Creation of an overlapping block structure.

• Multiplication of the block with an analysis window function (here, root-Hanning). This facilitates the use of a transformation into the frequency domain without so-called spectral leakage.

- Zero-padding the block with sample values to the desired block length for the transformation into the frequency domain.

5

- Transformation into the frequency domain, non-linear processing, back-transformation into the time domain

- Multiplying the block with a synthesis-window function
10   (here, root-Hanning) to reduce artifacts caused by the non-linear processing. The synthesis-window function hides this error at block boundaries and avoids audible discontinuities.

15 • Addition of overlapping blocks (overlap-add)

The WOLA is signal-transparent within itself, i.e., as long as no changes to the signal are made, the output signal corresponds to the input signal. The synthesis-window
20 function and the block overlap reduce unwanted signal changes, especially at block boundaries.

A detailed description can be found under point 9: Signal Processing.
25

3. Intensity Stereophony (IS)

In the case of pure intensity stereophony, a musical
30 instrument or a voice within the stereo base is mapped to a virtual spot by splitting the signal at a certain ratio to the left (L) and right (R) channel. The place is defined by

the relative levels of the left channel (L) and right channel (R). The signals in L and R are equal to each other in time/phase.

5   During reproduction, human hearing can determine the auditory event direction, and thus the origin of the sound source within the stereo base, by means of level differences between the left and right ear.

10   FIG. 4 shows the diagram of a signal division into the channels L and R in the case of pure IS.

The stereo base extends from the far left (R=0) to the center (L=R) up to the far right (L=0).

15

For FM stereo broadcasting, the audio signals L(t) and R(t) are matrixed.

A sum-signal $\sigma(t)$ and a difference signal $\delta(t)$ is created.
20   For L, R, σ and δ for the sake of simplicity, the time dependence is henceforth assumed and is omitted in the remaining representation.

The matrixing specification is as follows:

25

$\sigma = (L+R)/2$ and $\delta = (L-R)/2$

A de-matrixing takes place on the receiving side:

30     $L = \sigma + \delta$ and $R = \sigma - \delta$

First <u>a single sine wave signal</u> should be considered.

If we assume that on the transmitting end there is no excess width of the stereo base, in other words R=0 and L=0 represent the extreme locations of the stereo base, then results for

$$R = 0: \qquad \delta = \sigma$$

and for $L = 0:$ $\qquad \delta = -\sigma$

deriving: $\qquad |\delta| = |\sigma|$

For each spot mapped within the stereo base, regarding the sine wave signal follows

Rule 1 $\qquad |\delta| \leq |\sigma|$

where the equality holds true for the cases R=0 and L=0.

The absolute value function $|\sigma|$ can be regarded as an envelope, which is supported by relative maxima/minima of the sum signal.

FIG. 5 shows exemplary sum and difference signals as well as the resulting envelope curve (of a complex signal and not of a simple sinusoidal signal).

Rule 1 can be defined in the time and in the frequency domain of the audio signals:

a. Time domain: At any given time, the absolute value of
   the difference signal is smaller than that of the sum
   signal, or is at most equal to it. The difference
   signal lies within the envelope of the sum signal.

5   b. Frequency domain: At each frequency, the power of the
   difference signal is smaller than the power of the
   sum signal, or is at most equal to it.

Rule 1 leads to the following signal processing according
10   to the invention:

If the difference signal is superimposed with noise and the
above-mentioned rule is violated in the time domain or
frequency domain, then the difference signal can be reduced
15   in its absolute value to that of the sum signal at the
appropriate time resp. at the corresponding frequency.

The frequency spectrum with infinitesimal resolution is
defined as the sum of spectral lines. Each spectral line
20   can be interpreted as a vector with an amplitude value
(magnitude) and an associated phase value. A disturbance
can increase or decrease the amplitude value, change the
phase value, and leads to a corrupted channel separation
via the dematrixing.
25
The amplitude value is then reduced according to Rule 1b to
the value of the sum signal spectrum (corresponds to the
case R = 0 and L = 0, respectively). Because of the
infinitesimally small bandwidth, it is irrelevant whether
30   the power originates from the useful or interference signal
or both. The phase value is processed unchanged.

The signal processing is to be illustrated in three
examples according to FIG. 6 through FIG. 8:

As shown in FIG. 6, the sum signal is superimposed by the
5   difference signal such that the sum signal cannot be seen
in the manner shown in sections. The difference signal is
partly outside the envelope curve. A cut-off of the
interference signal components, and thus a reduction of the
disturbance in the time domain, is therefore possible.
10

According to the illustration in FIG. 7, however, a
reduction of the disturbance in the time domain is not
possible. Since the disturbed difference signal is still
within the envelope curve of the sum signal, no level
15   reduction can occur in the time domain.

In the frequency domain, a reduction of the disturbance is
possible, as shown in FIG. 8. The power of a frequency f in
the disturbed difference signal can be reduced to the power
20   of the corresponding frequency in the sum signal. The
extent of the reduction is indicated here as delta.

The application of rule 1 in the frequency domain
represents the largest share of noise reduction.
25

Often, many spectral lines of the difference signal are
lower than those of the sum signal, e.g., in the case of an
undisturbed signal in which the stereo effect is not
extremely pronounced (e.g., R = L / 2).
30

A reduction of amplitude values of the difference signal
spectrum is only carried out if the disturbance raises the

amplitude value above the value of the sum signal spectrum. This can occur particularly in the case of quiet passages, in which the disturbance dominates the useful signal.

5   With increasing frequency resolution (corresponding to increasing block length), more details of the spectra are opened, also in the form of gaps and sinks of the sum signal spectrum. The method exploits this situation and, in particular, lowers interfering signals in the difference
10   signal spectrum at these points.

It is relevant to human hearing how well this interference suppression works within the frequency groups. In a frequency group or critical bandwidth, the human hearing
15   evaluates the frequencies or spectral lines in common. There are 24 frequency groups from 0 to 20000 Hz.

FIG. 9 shows the critical band No. 9 from 920 Hz to 1080 Hz with 30 spectral lines, three of which are not lowered.
20
The above mentioned noise reduction at high frequency resolution causes an increase in the SNR within the frequency groups formed in the hearing.

25   The sum signal spectrum and the interference-reduced difference signal spectrum are mapped by the dematrixing into the channels L and R. In the frequency range there is a masking of possible residual errors of the interference suppression process. The masking depends on the statistical
30   properties and the spectral distribution of the useful signals in the left and right channels.

FIG. 10 shows the masking curves of a 1 kHz sinusoidal for different levels. If the 1 kHz tone has a level of 100 dB, for example, a 2 kHz tone with 70 dB cannot be perceived in the same channel.

5

It is understood that such masking effects in the frequency range also play a role for the perception of the processed signal, in particular with regard to residual errors, even if these masking effects do not enter the processing of the

10   signal.

The frequency groups are arranged in an approximately logarithmic frequency scale. For the purposes of the invention, for example a logarithmic scaling is also

15   conceivable in the transformation into the frequency domain. In the exemplary embodiment shown here, however, a linear scaling is performed.

According to the method of the invention, preferably no

20   spectral lines of the audio signal are combined into frequency groups. Rather, the evaluation of frequency groups is left to the human hearing, whereby the above-mentioned masking effects enter the perception of human hearing.

25

If we consider a composite signal instead of a single signal
then the situation is somewhat different.

30   FIG. 11a and 11b show an example:
L = sine signal 900 Hz, R = sine signal 300 Hz

FIG. 12 shows the sum signal and the difference signal (dotted)   .

5   Based on this example it can be seen that the difference signal can have a higher amplitude than the sum signal. Most cases are detected in more complex signal constellations with a factor of 1.4. It is also apparent that the maximum of the sum and difference signal does not

10   necessarily have to be concurrent. The envelope curve of the sum signal must be expanded by the factor k_{IS}, and relative maxima/minima of the sum signal must be held for a certain time so that the maxima/minima of the difference signal can be included. Since a maximum can occur first in

15   each of the two signals, the hold time should also apply to periods prior to the observation time.

The time difference between the extreme values of the sum signal and difference signal corresponds to a half period

20   of the higher frequency signal if the frequency ratio is 3:1. A time difference of +/- 3ms corresponds to 83/166 Hz. Frequencies in this range are usually mono, i.e., their share in the difference signal is low. A range of +/- 3ms covers nearly all such effects.

25

In intensity stereophonic signal constellations, for all mapped spots within the stereo base, the below rule applies:

modified Rule 1: $\quad |\delta| \leq k_{IS}|\sigma| \quad$ within a time window of τ_{IS}

with k_{IS} = amplitude factor in composite signals, for
example, 1.4

with k_{IS} = amplitude factor in single tones, for example,
1.1 *

with τ_{IS} = peak hold time, e.g., +/- 3 ms

The modified rule 1 is in the time domain and in the
frequency domain as follows:

a. <u>Time domain</u>: The difference signal is within an
envelope. The envelope is based on the relative
maxima/ minima of the sum signal, multiplied by a
factor of k_{IS}. Each newly detected and with k_{IS}
multiplied extreme value is kept within a time window
of τ_{IS} (peak hold).

b. <u>Frequency domain</u>: At each frequency, the power of the
difference signal is smaller than the power of the sum
signal, or is at most equal to it.

* This value (here, for example, 1.1) includes a level of
imbalance of the receiver of 1 dB between the left and
right channel. With of a factor k = 1.0 and a level
difference between the left and right audio output, the
difference signal would otherwise be cut unnecessarily.

This results in the following signal processing:

Time domain:

For each block, the disturbed difference signal is reduced to the envelope of the sum signal, wherein the envelope

5   takes into account signal shifts and amplitude increases. To compute the envelope, the relative maxima/minima (momentary peak values) of the absolute value of the sum signal are held (peak hold), and the resulting signal is increased by the factor k_{IS} (e.g., 1.4) for the range of

10   the time offset τ_{IS} (e.g. -3/+ 3 ms). It is necessary that the block length (in this case about 100 ms) covers the time difference between the extreme values of the sum and difference signals.

15   Frequency domain:

The amplitude value of each spectral line of the disturbed difference signal is reduced to the value of the sum signal. The phase spectrum of the difference signal is unchanged and processed further.

20

4. Time Based Stereophony and Intensity Stereophony with Time Based Stereophonic Fractions (LS)

25   With pure time based stereophony, a sound source is recorded with displaced microphones. The sound travels different distances to the microphones, depending on the input direction. Within the microphone signals L and R, signals are formed that have a direction-dependent time

30   delay. During reproduction, the human ear can determine the direction of the auditory event, and thereby locates the

sound by the time difference between the left and right ear signals.

FIG. 13 shows an example of such an AB microphone installation.

For the path length difference, $\Delta l = a*\sin\Theta$ applies for the time difference, $\Delta t = \Delta l/c$ applies, with c = 343 m/s , and the microphone distance a.

In practice, there is often no pure time-based stereophony, but an intensity stereophony with time-based stereophonic fractions, also referred to as equivalence stereophony. This manifests itself in different phase values in the spectra of the sum and difference signals, but also in non-simultaneous amplitude peaks in the time domain of both signals.

Recordings with time based stereophonic fractions are made for example in AB technology. The so-called Decca Tree is also used to reproduce complex sound bodies such as an orchestra, for example. Here, additional, laterally arranged supporting microphones are sometimes used. The sound from a source arrives at different microphones. Depending on the sound input direction and the arrangement of the microphones, individual levels and delay times are obtained for each microphone. The individual microphone signals are processed according to certain aspects, to a left and a right audio signal.

Depending on the sound input direction and the arrangement of the microphones, the following effects can be observed after matrixing:

5
- An auditory event, which is reflected in the difference signal, is not completely equal to the sum signal, depending on the input direction of the sound wave.

10
- The individual microphone signals are superimposed with their different time delays, and generate a statistical deviation of the amplitude of the difference signal compared with the sum signal. This is especially true for frequencies above the bass
15
range, in which the individual time delay results in ambiguousness of the phase (1m = 3ms = 360 degrees at 332 Hz!).

However, when recording, attention is paid to mono-
20
compatibility. That means, time delay differences are avoided between the microphone signals to prevent audible cancellation effects in the sum signal. Therefore, the intensity stereophonic portion in the signal predominates and the statistical amplitude distortion of the difference
25
signal is limited.

This leads to rule 2 of the method according to the invention:

Rule 2: $\qquad |\delta| \leq k_{LS}|\sigma|$ within the time window of τ_{LS}

with k_{LS} = amplitude factor, e.g., 1.4

with τ_{LS} = peak hold time, e.g., +/- 3 ms

Rule 2 is represented in the time and frequency domain as follows:

a. <u>Time domain</u>: The difference signal is within an envelope. The envelope is based on the relative maxima/minima of the sum signal, multiplied by a factor of k_{LS}. Each newly detected and with k_{LS}-multiplied extreme value is kept within a time window of τ_{LS} (peak hold).

b. <u>Frequency domain</u>: At each frequency, the power of the difference signal is smaller than the power of the sum signal, or is at most equal to it.

Rule 2 leads to the following processing of signals with time- based stereophonic fractions:

<u>Frequency domain</u>: Since time-shifted signals have the same magnitude spectra, the amplitude value (magnitude) of the disturbed spectral line of the difference signal can also be reduced in LS to the corresponding value of the sum signal spectrum. However, it is necessary that the block length (here, approximately 100 ms) covers the major time delay differences, i.e., that time-shifted signal components occur in the same block.

The phase spectrum is processed further, unchanged.

Time domain: For each block the disturbed difference signal
is reduced to the envelope of the sum signal, wherein the
envelope takes into account signal shifts and amplitude
increases (especially with transients). To compute the

5   envelope, relative maxima (momentary peak values) of the
absolute value of the sum signal are held (peak hold), and
the resulting signal is increased by the factor k_{LS} (e.g.,
1.4) for the time shift range τ_{LS} (e.g. -3/+ 3 ms).

10 **Rules 1 and 2** can also be applied to frequency groups in
the frequency domain. Thereby the powers of the individual
spectral lines are considered summarized.

Rule 1 is the consideration for intensity stereophony. Rule
15 1b or Rule 2b is always applied for the implementation of
the procedure.

5. Special Signal Constellations in LS

20 The method according to the invention also involves typical
signal constellations during signal processing in the case
of the time based stereophony:

25 - Cancellations

When recording signals with time-based stereophonic
fractions, frequency-selective cancellations (AL) can occur
in the sum and the difference signal spectrum. Due to
30 different signal path delays of the displaced microphones L
and R, a cancellation happens, e.g., in the sum signal σ =
(L+R)/2 if a frequency at microphone R undergoes a phase

rotation of 180 degrees with respect to microphone L. In the difference signal $\delta = (L-R)/2$, a cancellation happens if a frequency of the microphone R undergoes a delay-dependent phase rotation of 0 degrees. A cancellation or

5   destructive interference of sound waves in both signals at the same frequency can only occur if this frequency is derived from a variety of sound origins or input directions and both arrive at the microphones with equal amplitude. This is statistically unlikely. Usually, cancellations

10   occur in both signals at different frequencies.

FIG. 14 shows an undisturbed difference signal spectrum at LS. Frequency-selective cancellations in the sum signal spectrum at 2.09 kHz and 2.83 kHz can be seen.

15   Cancellations occur in both spectra at different frequencies.

A cancellation in the sum signal would decrease the difference signal spectrum severely at this frequency, in

20   accordance with rule 2b, and may impair the sound of the undisturbed audio signal. If a narrow cancellation is identified, a gain reduction can be avoided.

However, in the case of cancellations in the sum signal

25   spectrum, the unchanged value of the difference signal spectrum is not taken over, since this could be a pure interference signal of any magnitude. Instead, the value of the difference signal is reduced to the median value of the sum signal spectrum as shown in FIG. 15, showing a

30   disturbed difference signal spectrum. Thus, the noise reduction remains in effect without distorting the useful signal.

- Local level maxima

A local/frequency-selective maximum (LM) in the difference

5   signal spectrum can arise during the recording of time-
based stereophonic fractions by constructive interference
of sound waves, whereas the sum signal does not reach this
maximum. In this case, it would come to an undesirable
level attenuation in accordance with rule 2b.

10   To avoid a level reduction, there is a check to see if the
frequency-selective level maximum is associated with a
higher frequency-selective SNR. If so, the level remains
unchanged and is not lowered. Thereby it is assumed that
the noise has a white spectrum in the extended surrounding

15   of the level maximum and the frequency-selective level
maximum projects beyond this.
However, this strategy fails in the case of noise with a
frequency-selective spectrum. Therefore, the sum signal
spectrum is taken into account as an additional criterion.

20   Both the sum signal spectrum as well as the difference
signal spectrum must extend with their frequency-selective
maxima levels out of the spectral environment of the
difference signal spectrum. Then you can assume that the
frequency-selective level increase is resulting from the

25   useful and not from the interfering signal.

As can be seen in Fig
14, the sum signal spectrum and the difference signal
spectrum disturbed by noise have each two maxima at 2.12

30   kHz and 2.17 kHz, which rise above the values of the closer
environment (approximately 59 dB). It can therefore be
assumed that both maxima result from the useful signal, and

that the local SNR is high. These maxima of the difference
signal spectrum may remain unchanged for further signal
processing.

5   FIG. 16 shows a frequency-selective noise spectrum.
However, the difference signal spectrum shows several
spectral lines with a higher level that is not supported by
the spectral lines of the sum signal. It can be deduced,
therefore, that the high-level spectral lines of the
10   difference signal originate from an interference signal. A
reduction to the level of the sum signal spectrum can be
carried out.

Local maxima are identified separately for the sum signal
15   and the difference signal spectrum. The median-filtered
difference signal spectrum can be seen in FIG. 17a. A
maximum in one of the spectra exists when the spectrum
exceeds its median value for a specified value in dB.

20   In FIGs 17a and 17b, LMSumme and LMDifferenz (labels in the
bottom of the screen) indicate the maxima of the sum signal
spectrum or difference signal spectrum. If LMSumme lies
within the bandwidth of LMDifferenz, a superordinate local
maximum LM is reported, which causes a level reduction of
25   the difference signal spectrum in the corresponding
frequency bandwidth to be blocked. In the case of the
example in Fig. 17b, this is true only for a narrow
frequency band at 5.75 kHz.

30   Both cancellations (AL) and local maxima (LM) are
identified by means of median filtering. Both cases are
incorporated into the spectral correction function as non-

linear signal processing. LM and AL contribute to the restoration of the undisturbed difference signal spectrum.

6. Identification of IS and LS

The identification of LS takes place via a cross-correlation (similarity analysis) of the sum and difference signals. The basic idea behind this is that the time difference of both signals is determined by the cross-correlation. If this is equal to zero, IS is present; otherwise, LS is present.

Complementarily or alternatively, an identification can take place in the frequency domain if cancellations occur in the sum and difference signals at different frequencies.

The cross-correlation function (KKF) is calculated from one block each of the sum signal and difference signal. In almost all recordings, including those with time-based stereophonic character, the low tones are monaural. They produce small phase differences at the different microphones, and generally dominate higher frequencies in the level. Time delays at higher frequencies are covered and not recognized by the KKF. In order to avoid this, the sum and difference signals are initially differentiated in time and only then is the KKF calculated. The differentiation in the time domain corresponds to an increase in the level to higher frequencies in the frequency domain.

The KKF is calculated independently of the level by determining the covariance function (KOV) (for formulas see

appendix). Maxima can be identified by subsequent
calculation of the absolute value independently of the
signal polarity. Maxima are shown at such time shifts, in
which the differentiated sum and difference signals show

5   similarities. In the case of pure IS, the maximum shows up
at the time shift zero. More KOV maxima may occur when both
signals have inner similarities: the difference signal is
often an attenuated copy of the sum signal. To hide these
maxima, the autocovariance (AKOV) of the sum signal is

10   calculated and subtracted from the scaled to 1 KOV
covariance. If the difference (delta KOV) exceeds a certain
threshold value, LS is present; otherwise, IS is present.

FIGs. 18a and 18b show typical delta KOV at LS (18a) and IS

15   (18b).

In the case of superimposed disturbances, the above-
mentioned method does not provide reliable detection of
time-based stereophonic delays between sum and difference

20   signals in each block. Disturbances change the time course
of the difference signal, lead to dissimilarity of sum and
difference signals, and consequently reduce the level of
delta. Thus, in order to avoid the occurrence of
disturbance-related IS/LS fault decisions, the signal-to-

25   noise ratio (SNR) is considered block by block. The SNR is
here defined as the ratio of the power of the sum signal
and difference signal. Should the SNR decrease below a
threshold value in a block (making the IS/LS-decision
unreliable), the decision of the last block with a high SNR

30   is used. The starting value LS is predetermined.

7. Signal Classification and Temporal Processing

Audio signals can have a transient or stationary character.
Transient signals are characterized by an increase in power
within the shortest periods of time, often associated with
preceding signal pauses or silent passages. Stationary
signals have a more continuous power timeline.

Interference can most effectively be reduced in the
frequency domain if the magnitude spectra of the useful
signal and that of the interfering signal differ
significantly.

Unfortunately this does not apply to transient useful
signals (such as guiro and castanets) because they have an
almost white spectrum and there is little difference to the
noise as interfering signal. A reduction in selective
frequencies can hardly take place. The residual noise is
therefore high in such cases. Further disturbances are
added: The processing of noise in the frequency domain and
transformation back into the time domain causes an alias,
spread out over the block. It is mostly hidden for
stationary useful signals.
As long as the transient useful signal is present, residual
noise is masked simultaneously. Residual noise that occurs
after a transient can be better masked, because the natural
transient signals settle more slowly, and hearing has
temporal post-masking. The masking of residual noise, which
precedes a transient, is lower. In signal pauses before a
transient noise, a so-called pre-echo can be audible.

FIG. 19 shows the masking in the time domain ("temporal masking").

If the useful signal has a transient/impulse character and
noise is superimposed on the difference signal, then the
residual noise can be reduced (the pre-echo amongst
others), by an additional signal processing in the time
domain (temporal processing). The difference signal is
herewith limited to the envelope of the sum signal
(clipping).

The following images using an example of a transient signal
show how a temporal processing reduces pre-echoes.

FIG. 20 shows an undisturbed guiro without signal
processing, i.e., the original signal.

FIG. 21 shows a noisy guiro after signal processing in the
frequency domain. A pre-echo is present.

FIG. 22 shows the noisy guiro after signal processing in
the time and frequency domain. The pre-echo is
significantly reduced.

If the useful signal is transient and/or short signal
pauses are present within the block, then clipping can also
help to temporally eliminate or reduce disturbances with a
transient character.

The limiting to the envelope curve reduces the interfering
energy in the case of strong disturbances. In these cases,
the magnitude spectrum of the difference signal after

clipping lies below that of the untreated magnitude
spectrum. The effect of the original disturbance on the
magnitude and phase of spectral lines is reduced.
On the other hand, the clipping itself produces noise with
5   a white spectrum, which manifests itself as interferent
portions in the magnitude and the phase of the difference
signal spectrum. This effect increases the more signal
components are cut off. If it comes to a level increase in
spectral lines in this regard, this can be corrected by the
10   signal processing in the frequency domain. Level reductions
cannot be corrected.
The distorted phase spectrum is taken on unchanged.

It is therefore necessary to decide from block to block
15   whether the limiting to the envelope curve is to be
applied.

8. Criterion for the Use of Temporal Processing

20   If the sum signal (useful signal) is stationary, or if it
has a temporally continuous signal form, its frequency
spectrum usually provides sufficient gaps for an effective
selective interference reduction in the frequency domain of
the difference signal. In the case of stationary signals,
25   the temporal processing (limiting to the envelope curve)
worsens the residual noise and thus also the channel
separation. Therefore it is better to turn off temporal
processing in this case.

30   In contrast, if the transients dominate or signal pauses
exist within the signal, it is advantageous to additionally
use temporal processing. In these instances, the temporal

processing reduces the pre-echoes, in particular. Pre-echoes occur as a form of alias after the IFFT, are distinguishable in the signal pause before a transient, and may be audible without temporal processing. In the case of

5   stationary signals, the alias after the IFFT is usually masked by the continuous signal form.

From this follows Rule 3:

10

Rule 3: Temporal processing (limiting to the envelope curve) is turned on, if in the sum signal (useful signal) transients dominate or when there are pauses within the signal.

15

It is useful to already check for the condition within the time domain, since a limiting to the envelope curve is, before the signal processing, in the frequency domain.

20   For identification of transients, the envelope of the sum signal σ is examined. A transient is considered identified if the envelope curve increases by more than x percent within a time segment Δt. Determination in percentages allows for level-independent identification.

25

9. Signal Processing

9.1. Signal Processing in the Time Domain, part 1
30   (The sequence of steps 1 and 2 is interchangeable).

1. Block forming of the audio sample values for the right and left channel. An overlapping block structure is produced. The overlap is, e.g., 50%. The block length is, e.g., 4096. The following processing steps apply per block.

2. Matrixing the channels L and R into

   $\sigma = (L + R) / 2$ and $\delta = (L - R) / 2$

   (alternatively direct processing of the sum signal σ and difference signal δ)

3. Signal analysis and limiting to the envelope

   3.1 Identification of LS or IS on the basis of σ and δ

       3.1.1   Temporal derivative of σ and δ

                  Blocks $d\sigma$ and $d\delta$ are created

       3.1.2   Calculation of the 1-normalized absolute value of the covariance of $d\sigma$ and $d\delta$: absKOVnorm

       3.1.3   Calculation of the absolute value of the autocovariance of $d\sigma$: absAKOV

       3.1.4   Calculation of the difference deltaKOV = absKOVnorm - absAKOV

       3.1.5   Temporal limitation of deltaKOV to an upper   limit (here, 3 ms): deltaKOVlim is generated

       3.1.6   Calculation of the maximum of deltaKOVlim

       3.1.7   Identification of LS or IS:

                    Verification of SNR:

                    if $rootSNR < rootSNR_{thresh}$ (or if $SNR < SNR_{thresh}$ {e.g. 0.3}):

the LS/IS-decision of the

previous block is adopted

else:

if max (deltaKOV) <kovlevel

5 (e.g. kovlevel<0.1):

IS

else: LS

10 3.1.8 Calculation of the envelope of σ

regarding the values of the time shift

τ and of the amplitude factor Ampf, e.g.:

LS: τ_{LS} = +/-3.0 ms k_{LS} = 1.4

IS: τ_{IS} = +/-3.0 ms k_{IS} = 1.4

15

3.1.9 Identification of transients and limiting

of δ:

Calculation of the percentage increase

20 (PA) of the envelope of σ within a time

interval of n samples.

if PA <x%: stationary signal

else: detected transient, limiting

25 the signal δ to the envelope

of σ

3.2 Weighting of each block with an analysis window

function (here, root-Hanning):

30 The weighted blocks wσ and wδ are created.

9.2 Signal Processing in the Frequency Domain

4. Zero-padding ** of the weighted blocks wσ and wδ.

5. Transformation into the frequency domain. Outcome is the spectra $W\Sigma(f)$ and $W\Delta(f)$.

6. Separation into magnitude and phase spectra.

7. Calculation of the spectral correction function K(f) (see annex).

8. Multiplication of the magnitude spectrum of $W\Delta(f)$ by correcting function K(f) in the linear measure.

9. Calculation of the corrected complex spectrum of $W\Delta(f)$.

10. Inverse transformation into the time domain. Outcome is a corrected difference signal δ(t).

** Padding the block with zeros to the desired length (power of 2 for FFT)

9.3 Signal Processing in the Time Domain, part 2

11. Multiplication of the block with a synthesis window function (here, root-Hanning).

12. Overlap add of blocks

13. Dematrixing of σ and the corrected δ into the channels L and R

Annex A / Formulas:

Hanning window (analysis and synthesis window function)

5

H (n, N) = 0.5 - 0.5cos {(2 Π n / (N-1)} where N = number of samples per block

root Hanning = $\sqrt{}$ H (n, N)

10

Signal to Noise ratio SNR within a block:

SNR = $P_{w\sigma}$ / $P_{w\delta}$ with P = power = rootSNR =$\sqrt{}$ SNR

15

Covariance KOV and autocovariance AKOV:

20

Mean value

$$m(x) := \frac{1}{n} \cdot \sum_{i=0}^{n} x_i$$

25 Variance

$$var(x) := \sum_{i=0}^{i_{max}} \frac{(x - m(x))^2}{i_{max} - 1}$$

30 Standard deviation

$$stdev(x) := \sqrt{var(x)}$$

Covariance

$$kov(\sigma,\delta)_i := \sum_k \left[\left(\sigma_k - m(\sigma) \right) \cdot \left(\delta_{i+k} - m(\delta) \right) \right]$$

5

Normalized covariance

$$KOV(\sigma,\delta)_i := \frac{kov(\sigma,\delta)_i}{i_{max} \cdot stdev(\sigma) \cdot stdev(\delta)}$$

10

Autocovariance

$$AKOV(x)_i := KOV(x,x)_i$$

15

20   **spectral correction function K (f) considering the spectra
on a linear scale:**

local level maxima in the difference signal spectrum:

25

For LS:

Calculating the median filtered sum signal spectrum
$W\Sigma(f)_{median}$ and difference signal spectrum $W\Delta(f)_{median}$

30

Determining the frequencies $f_{LM\ Diff}$ with local level
maxima ($>$ LM_{Diff} dB) in $W\Delta(f)$

Determining the frequencies $f_{LM\ sum}$ with local level
5   maxima ($>$ LM_{sum} dB) in $W\Sigma(f)$

if in a contiguous range of $f_{LM\ Diff}$, a frequency $f_{LM\ sum}$
occurs, then
10

$$K(f) = 1$$

otherwise:

if $|W\Delta(f)| > |W\Sigma(f)|$: $K(f) = |W\Sigma(f)| / |W\Delta(f)|$
15

if $|W\Delta(f)| \leq |W\Sigma(f)|$: $K(f) = 1$

for IS:

20   if $|W\Delta(f)| > |W\Sigma(f)|$: $K(f) = |W\Sigma(f)| / |W\Delta(f)|$

if $|W\Delta(f)| \leq |W\Sigma(f)|$: $K(f) = 1$

25

Cancellations in the sum signal spectrum:

30   For LS:

Calculating the median filtered sum signal spectrum $W\Sigma(f)_{median}$

Identifying the frequencies f_{AL}, for which time-delay-dependent cancellations (narrow dips) take place in $W\Sigma(f)$.

if $f = f_{AL}$ $K(f) = W\Sigma(f)_{median}$

else f:

if $|W\Delta(f)| > |W\Sigma(f)|$: $K(f) = |W\Sigma(f)| / |W\Delta(f)|$

if $|W\Delta(f)| \leq |W\Sigma(f)|$: $K(f) = 1$

For IS:

if $|W\Delta(f)| > |W\Sigma(f)|$: $K(f) = |W\Sigma(f)| / |W\Delta(f)|$

if $|W\Delta(f)| \leq |W\Sigma(f)|$: $K(f) = 1$

The method according to the invention reduces noise and other types of interference that occur in the difference signal and disturbances that exceed the sum signal. Interferences include those caused by the transmission chain after matrixing in the stereo coder up to the FM demodulator in the receiver, e.g., inherent noise of the FM transmitter; radio transmission interference; noise due to

low power of the receiving antenna; the inherent noise in the RF part of the receiver; RF-adjacent channel and co-channel interference; quantization noise of the ADCs in the IF-range of the receiver; non-linear distortion products

5   due to the limitation of the IF bandwidth (as long as they are not within the sum channel); disturbances due to signals of purely digital or hybrid transmission systems, such as IBOC, HD Radio and FMeXtra; and disturbances and crosstalk within hybrid systems, which have an impact on

10   the difference signal of the analog transmission system.

Interference that occurs in the sum channel, i.e., also in the case of pure mono-reception, cannot be eliminated by this method. This includes adjacent channel interference, which can particularly cause brief but strong disturbances

15   during mobile reception.

The improvements mentioned also refer to the applied FM variant SSBSC in the USA. The invention approach is fully compatible with SSBSC.

20   The method was simulated and emulated in this exemplary embodiment with a mathematical program on a PC.

Referring to the block diagrams according to FIGS. 23a to 25, an exemplary embodiment of the method according to the

25   invention is to be explained.

As shown in FIG. 23a, the analog FM stereo signal is first digitized and matrixed. For a receiver with digital signal processing, the already-digitized signal can be used.

30

The signal is divided into a sum signal and a difference
signal in the time domain, and weighted, overlapping blocks
are generated. The weighting can, for example, be carried
out using the Hanning function (window function).

5

The sum signal is used both for the calculation of the
envelope curve and for the identification of time-based
stereophony (LS) and intensity stereophony (IS).

10   The identification of LS and IS preferably takes place, as
previously described, by means of a correlation analysis.

The difference signal can be lowered to the envelope curve
of the sum signal.

15

For this purpose, according to a preferred embodiment of
the invention, a transient detection is provided, which
decides whether it is a transient or stationary signal. In
the case of a stationary signal, a limiting to the envelope
20   is not performed and the unchanged difference signal is
used immediately.

The transient signal, on the other hand, is subjected to
the limitation to the envelope curve of the sum signal.

25

Then, both the sum signal and the difference signal are
transformed from the time domain into the frequency domain.

The frequency range, or the part of the method in which the
30   processing takes place in the frequency domain, is marked
in this block circuit diagram in the dotted frame, which is
characterized by frequency range.

There is now a sum signal spectrum which has an amount as well as a difference signal spectrum, which also has an amount.

5

The phase of the difference signal spectrum is processed further unchanged.

As shown in FIG. 23b, cancellations at certain frequencies
10   are identified over the sum signal spectrum.

The affected frequencies or spectral lines can be defined by means of an identification and comparison of the maxima of the sum signal spectrum and the difference signal
15   spectrum.

At cancellations, the difference signal spectrum is reduced to the median value of the sum signal spectrum.

20   The identification of cancellations is explained below with reference to FIG. 24.

Identification and comparison of the maxima is explained below with reference to FIG. 25.
25

In the case of intensity stereophony, the difference signal spectrum is directly processed further without the need for identifying cancellations and local maxima, or the need to use the identification process in this exemplary
30   embodiment.

For all frequencies that are not identified, rule 1 is
executed in the frequency domain and the difference signal
spectrum is reduced to the sum signal spectrum.

5   A corrected difference signal spectrum is generated. This
is transformed back into the time domain using the phase
spectrum.

After weighting and combining the overlapping blocks in the
10   time domain, a corrected difference signal is generated.

The sum signal and the corrected difference signal are
dematrixed and a corrected stereo signal is generated.

15   FIG. 24 shows an embodiment of the identification of
cancellations shown as a block in FIG. 23b.

The logarithmized sum signal spectrum is compared with its
median curve. If the difference is above a threshold value,
20   the difference signal spectrum is reduced to the respective
median value. If not, rule 1 applies and the difference
signal spectrum is reduced to the sum signal spectrum as
shown in FIG. 23b.

25   FIG. 25 shows the identification of the maxima depicted in
FIG. 23b.

Both the logarithmized sum signal spectrum and the
logarithmised difference signal spectrum are subjected to a
30   median filtering.

If the respective difference is above a threshold value, a maximum can be identified.

If a maximum of the sum signal spectrum is within the
5   frequency bandwidth of the maximum of the difference signal spectrum, the difference signal spectrum is not lowered for this frequency bandwidth.

By means of the invention, a reduction of disturbances of a
10   stereo signal can take place, so that this reaches approximately the quality of the monosignal.

Claims:

1. A method for processing an analog FM stereo signal, comprising:
- digitizing the analog FM stereo signal as a sum and difference signal;
- dividing the digitized signal into overlapping blocks;
- transforming the overlapping blocks into the frequency domain;
- comparing the spectral lines of the difference signal spectrum with the spectral lines of the sum signal spectrum;
- attenuating at least spectral lines of the difference signal spectrum, if these, in each case, have a higher magnitude than the respective spectral line of the sum signal spectrum; and
- transforming-back the sum and difference signal spectrum and merging the overlapping blocks.

2. The method according to claim 1, wherein, if the width of a relative minimum of the spectrum of the sum signal is below a threshold value and the depth exceeds a threshold value and there is thus a cancellation, the spectral lines of the difference signal are not attenuated or less attentuated.

3. The method according to claim 1 or 2, wherein, in a range of a local maximum of the difference signal spectrum, this is compared with the sum signal spectrum and the difference signal spectrum is not attenuated in this range if a maximum of the sum channel spectrum lies within the frequency bandwidth of the maximum of the difference signal spectrum.

4. The method according to claim 2 or 3, wherein differences in a time between sum and difference signals or between parts of both signals are determined by means of a similarity analysis or bymeans of a correlation in the time domain and/or frequency domain, and wherein the method steps are carried out in case of differences in time.

5. The method according to any one of claims 2 to 4, wherein, in case of a cancellation of a spectral line of the sum signal spectrum, the spectral line of the difference signal spectrum is only attenuated to a spectral substitute value.

6. The method according to any one of claims 1 to 5, wherein the difference signal in the time domain is limited to an envelope curve of the sum signal whose maxima/minima are held, defined as a peak hold, for a period of time before and after the peak entry time, and a thus-changed envelope curve or the sum signal is multiplied by a factor greater than 1, including a factor between 1 and 2.

7. The method according to claim 6, wherein limitation of the difference signal in the time domain to the envelope curve of the sum signal is only performed when a transient character of the signal is detected via an evaluation of the sum signal or its envelope curve.

8. A method according to any one of claims 1 to 7, wherein an FM stereo signal is digitalized, said digitalized signal is subjected to a signal processing, wherein the digital signal processing is performed such that the FM stereo signal is viewed so that its signal portions can only be located within the stereo base.

9. A method according to any one of claims 1 to 8, wherein the spectral lines of the difference signal spectrum are attenuated to the magnitude of the respective spectral line of the sum signal spectrum.

10. A computer program product comprising a computer readable medium storing thereon a plurality of computer executable instructions that when executed by one of a computer,a smartphone, and consumer electronics device, the instructions, when processed, perform a method according to any one of claims 1 to 9.

11. Consumer electronics device or smartphone, comprising means for carrying out a method according to any one of claims 1 to 9.

12. Consumer electronics device or smartphone according to claim 11, wherein the means for carrying out the method comprise an application-specific integrated circuit (ASIC) or a logical circuit programmed according to any one of claims 1 to 9, or a data memory comprising a plurality of instructions, the instructions when executed by a microprocessor or microcontroller performing a method according to any one of claims 1 to 9.
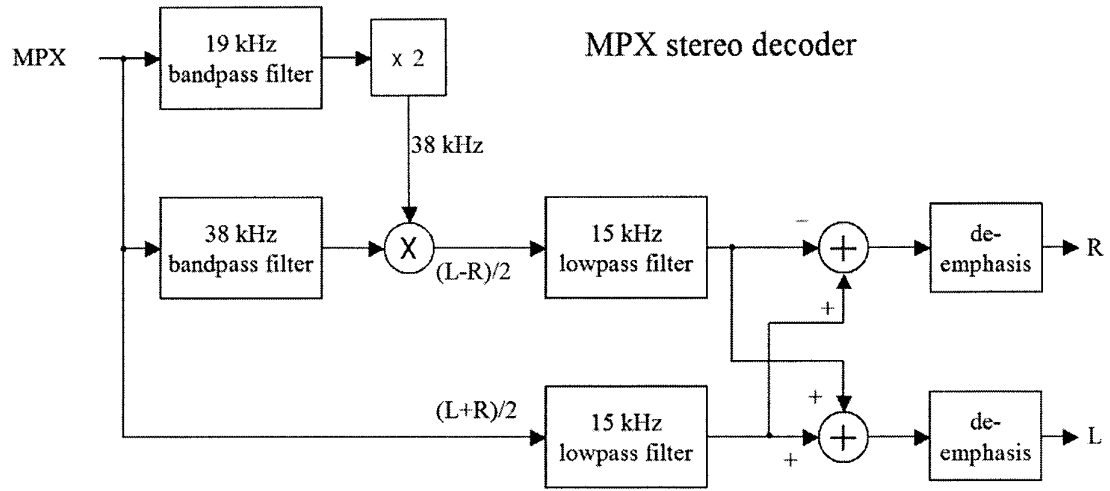
Fig. 1 (prior art)
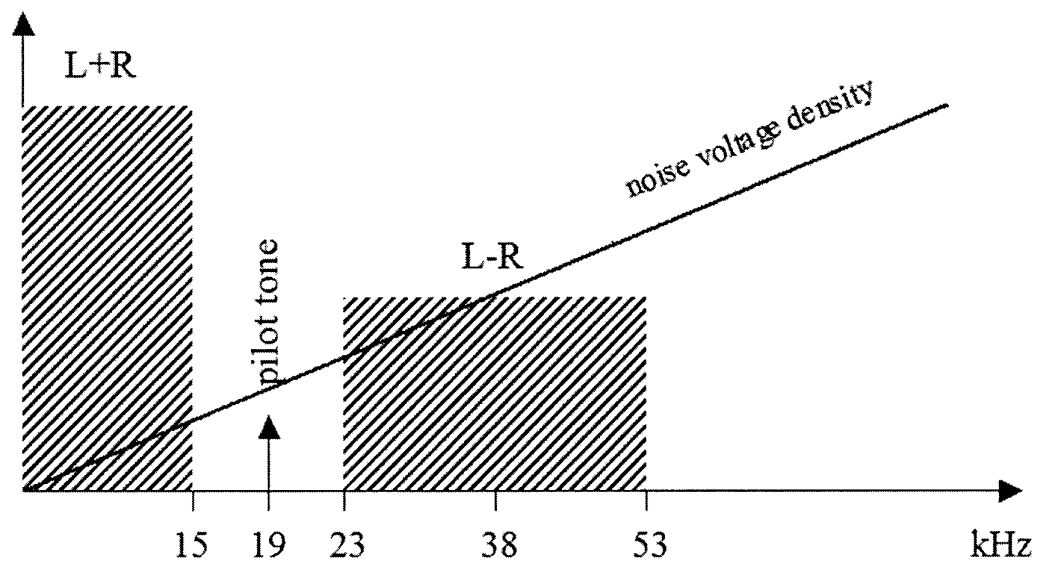
Fig. 2 (prior art)
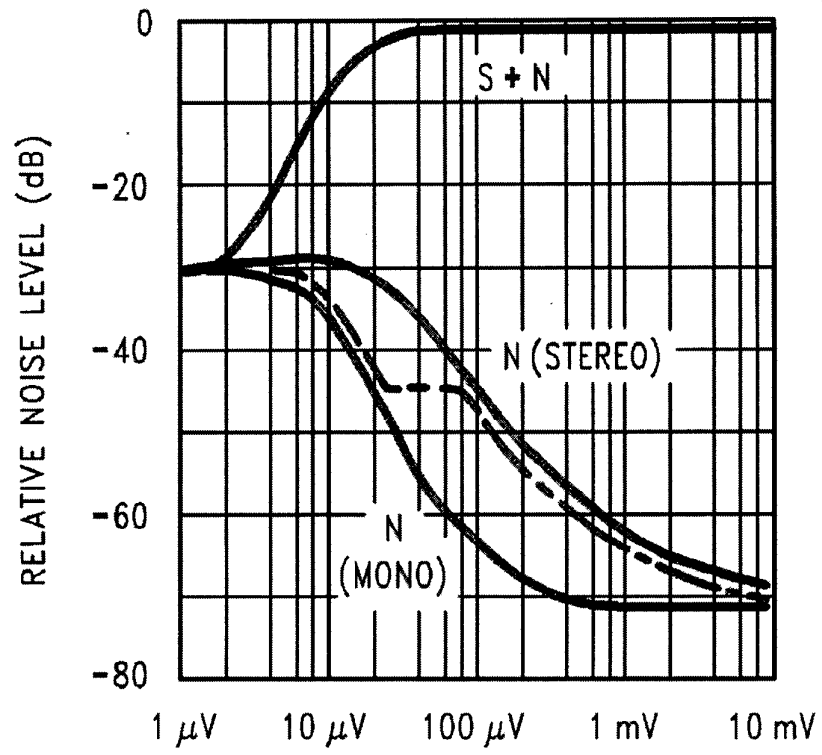
Fig. 3 (prior art)
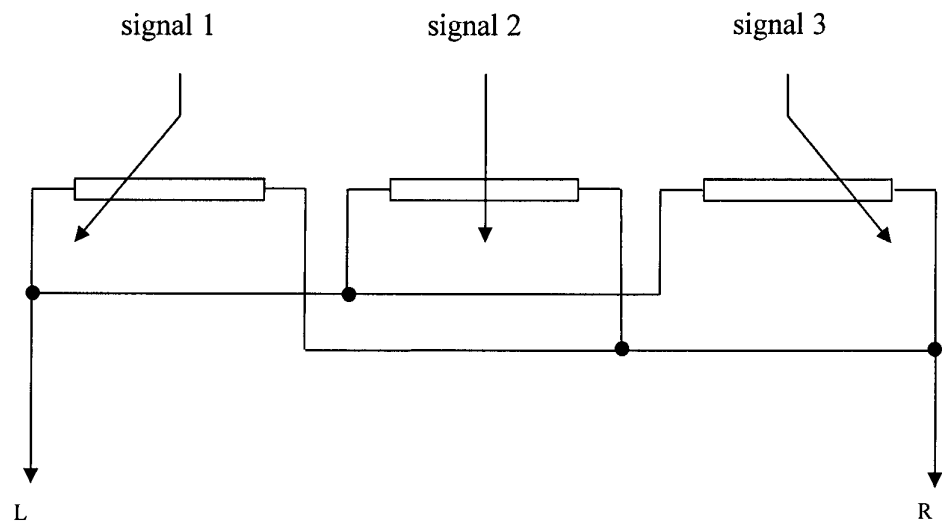
Fig. 4 (prior art)
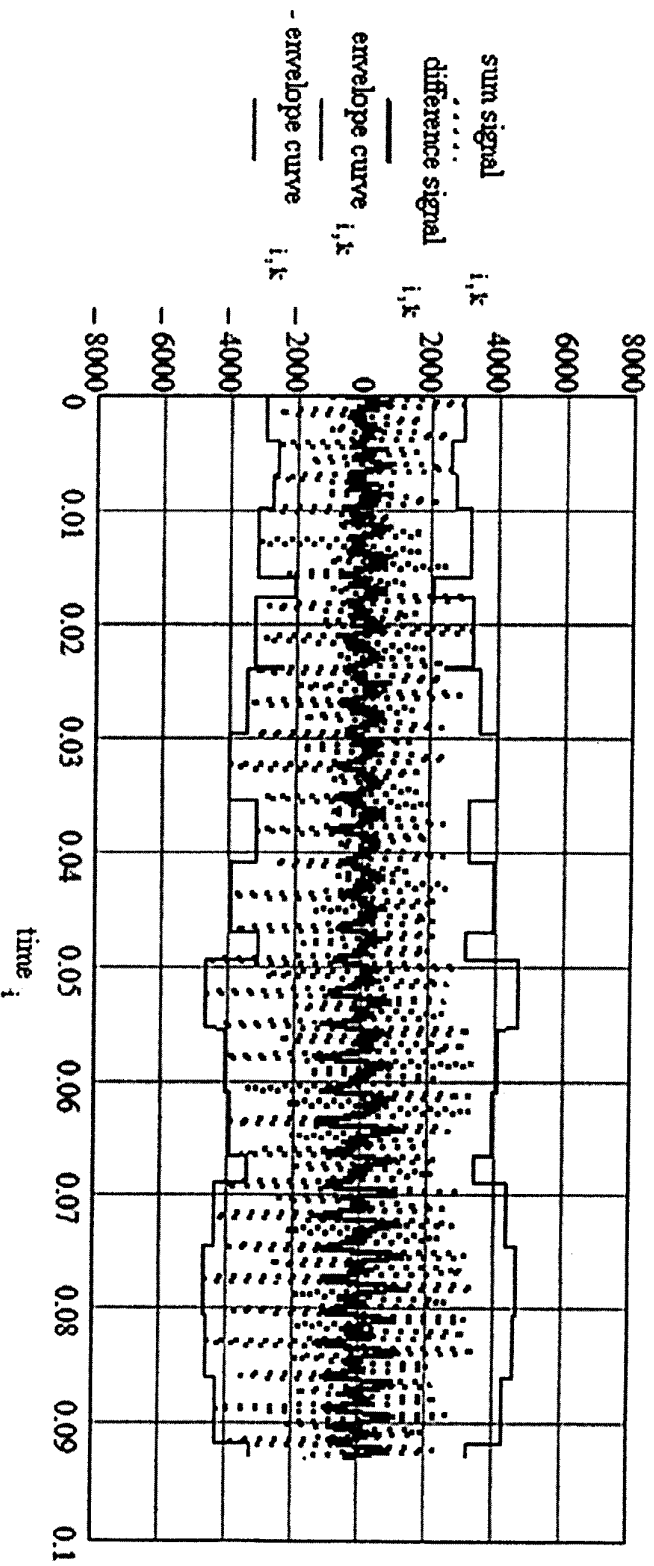
Fig. 5
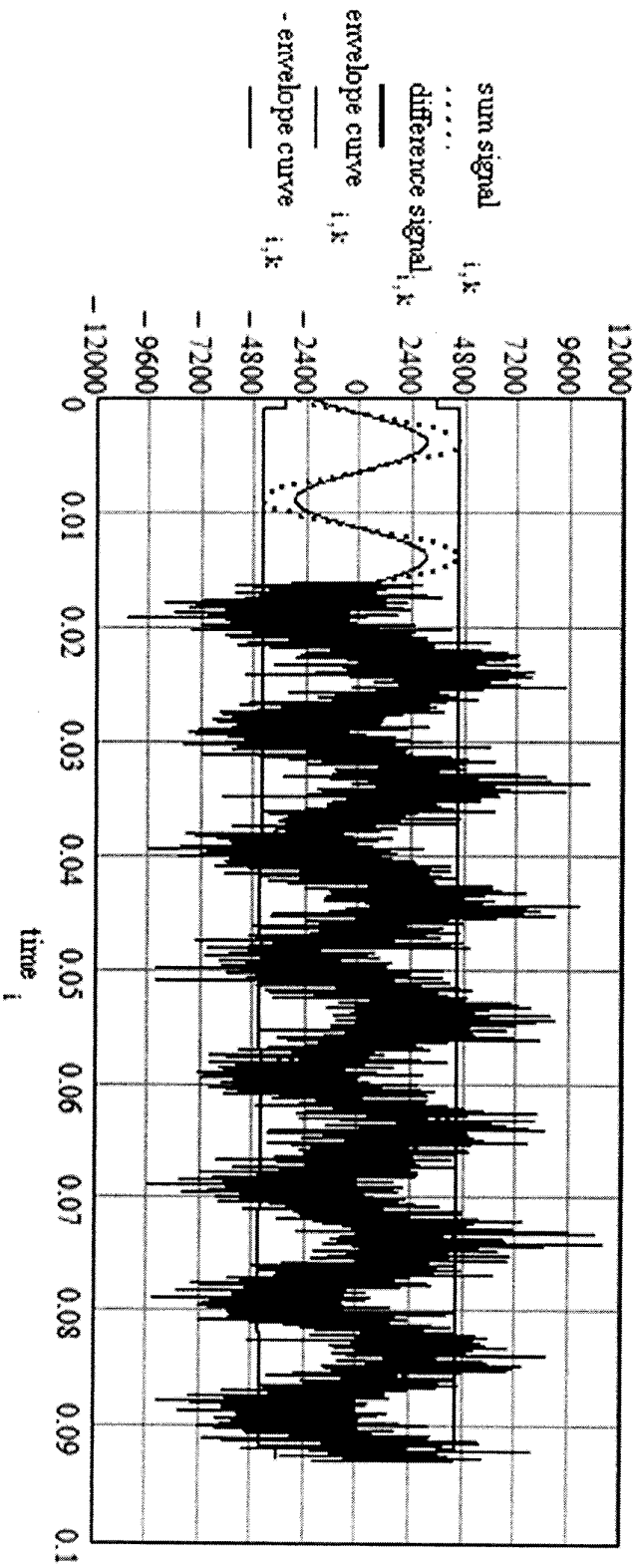
Fig. 6
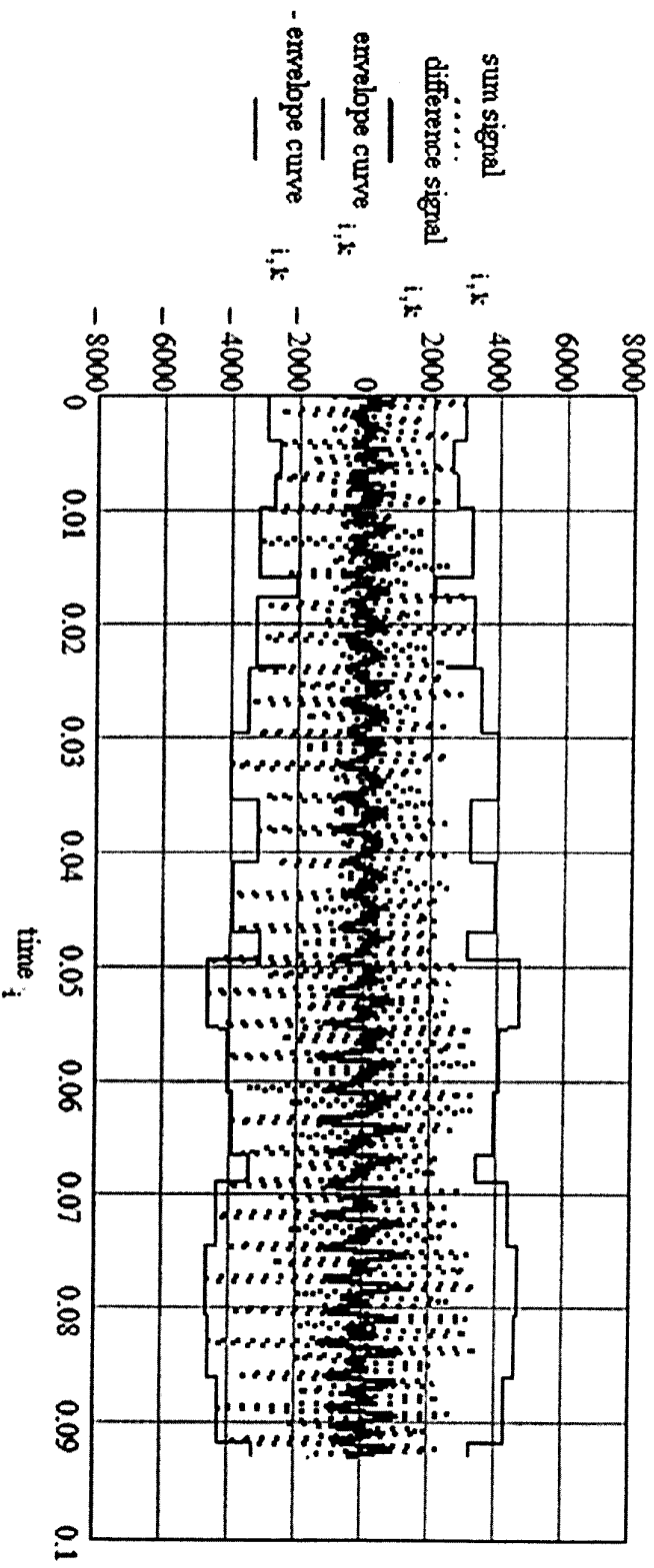
Fig. 7
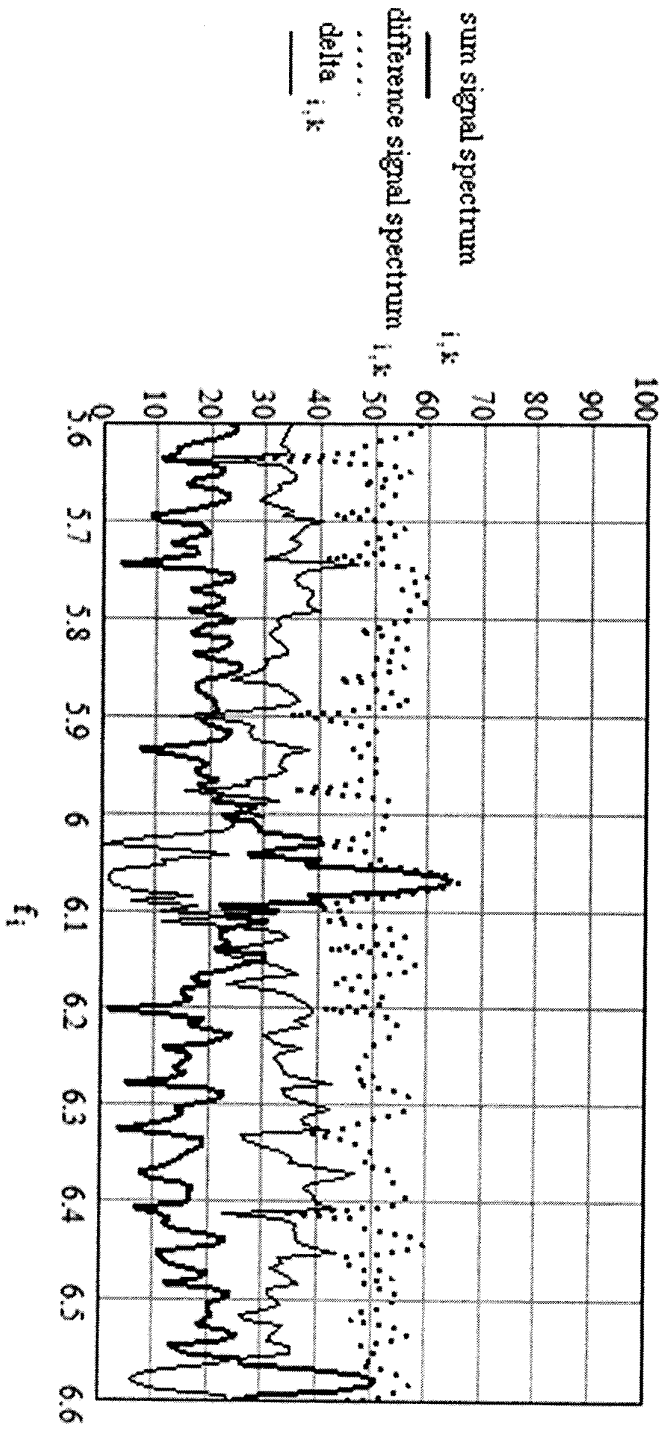
Fig. 8
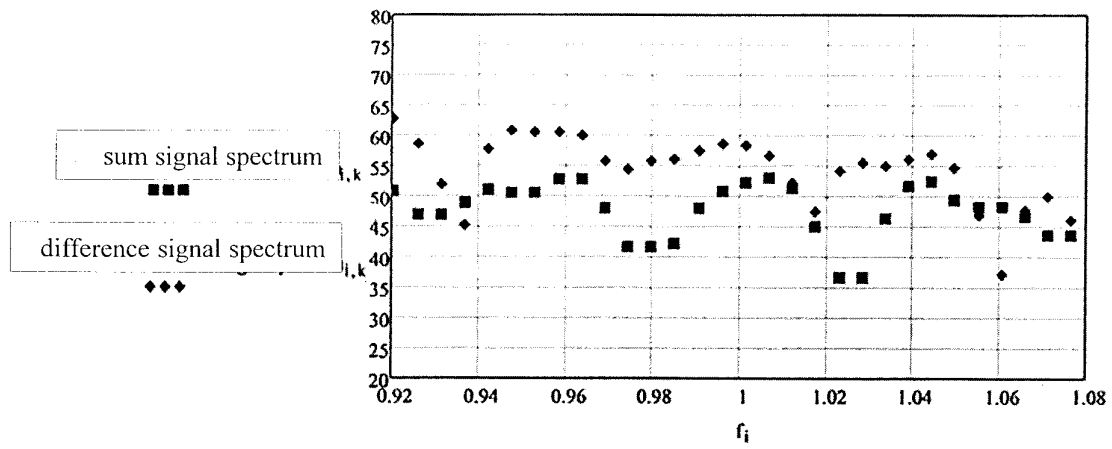
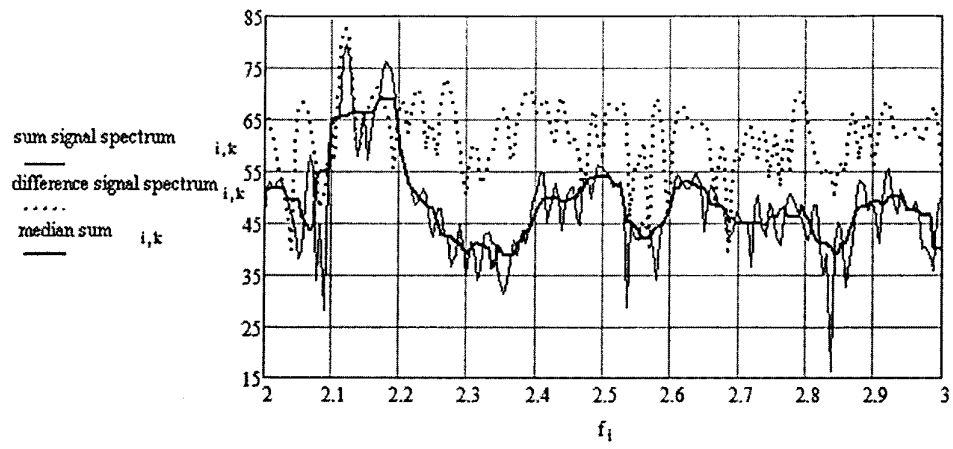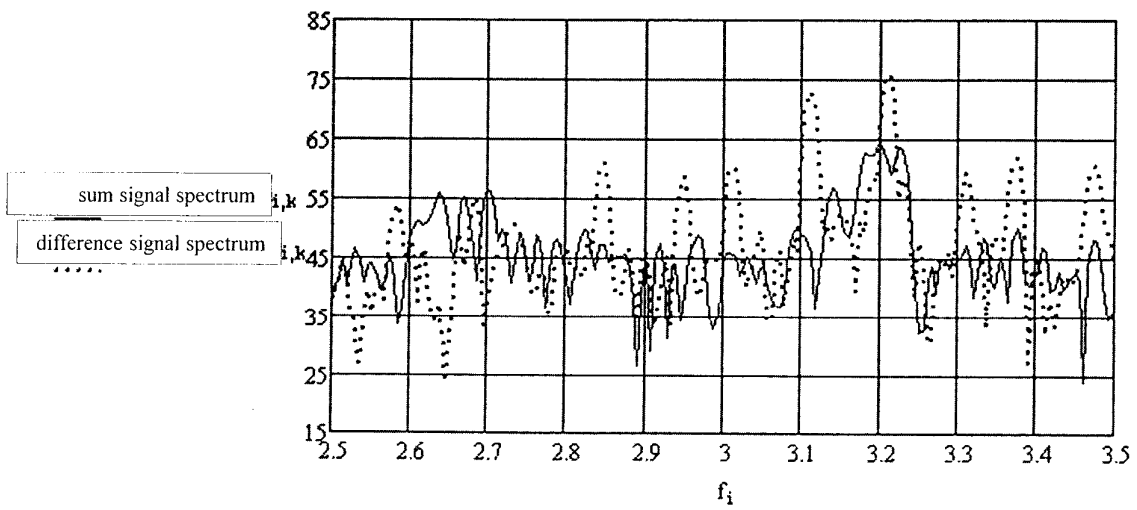Fig. 9
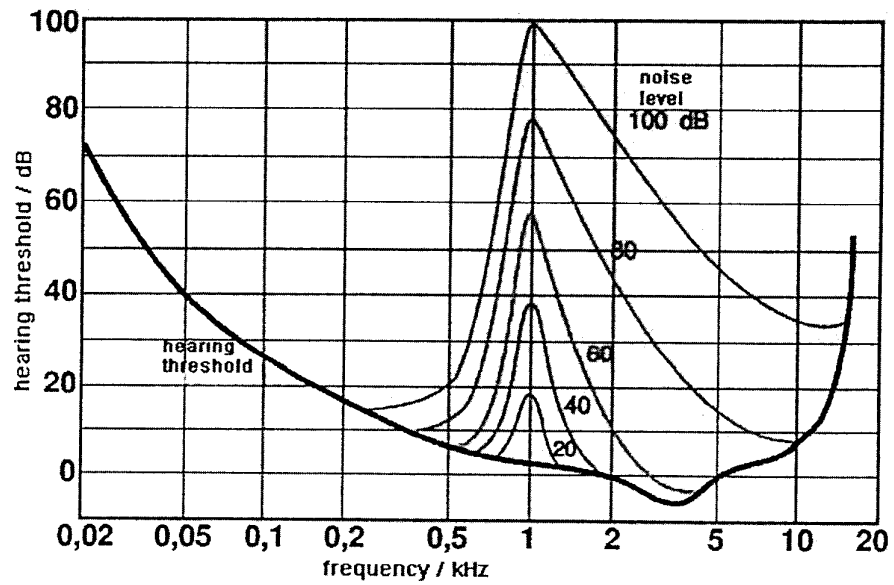
Fig. 10
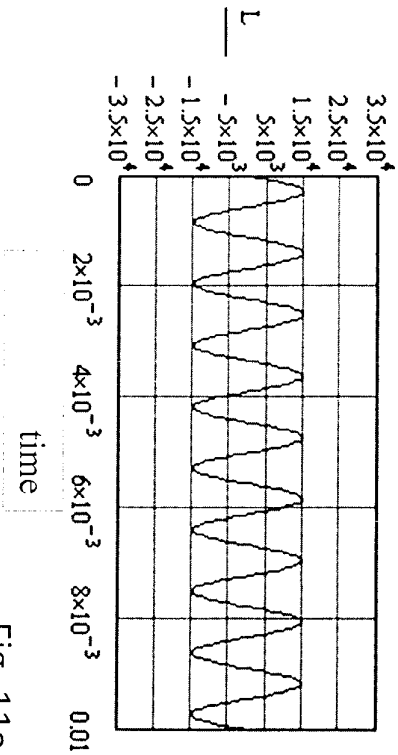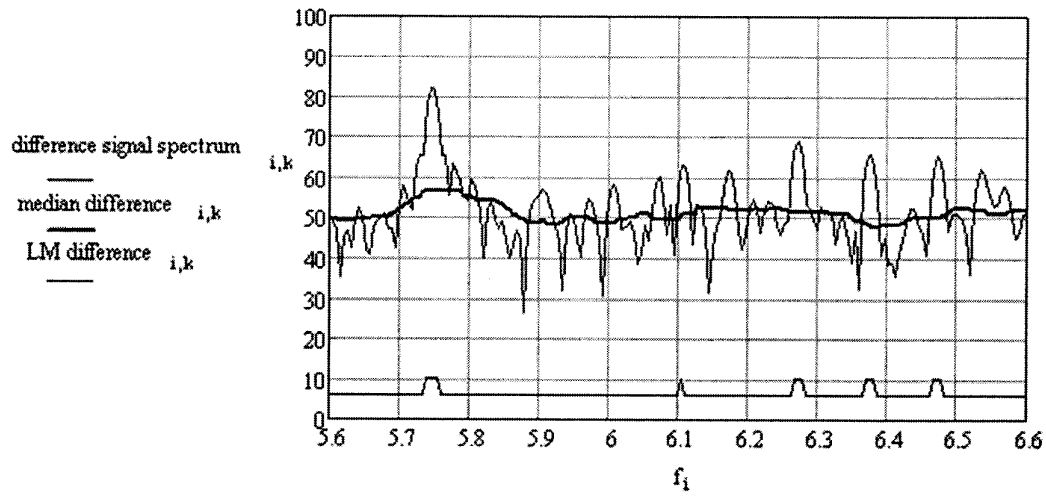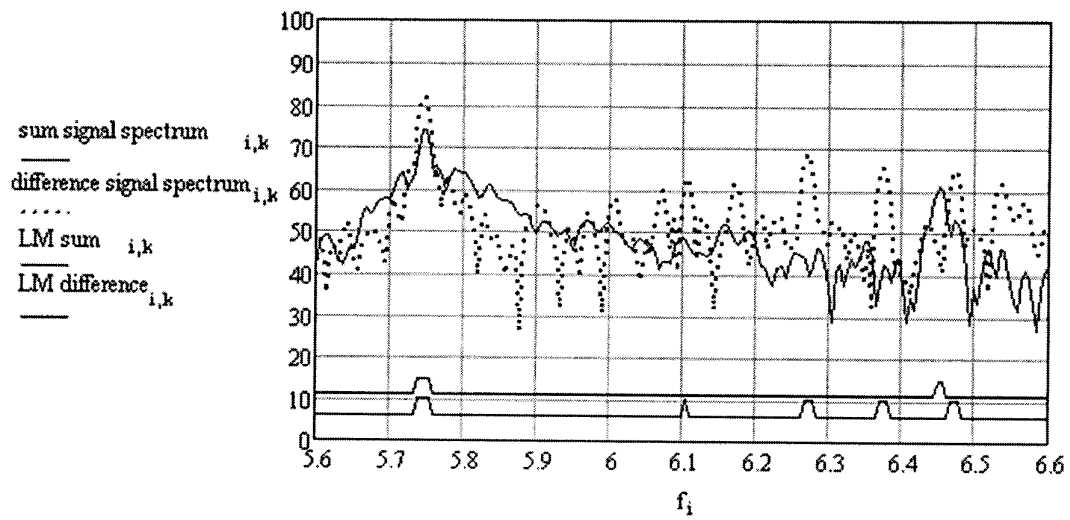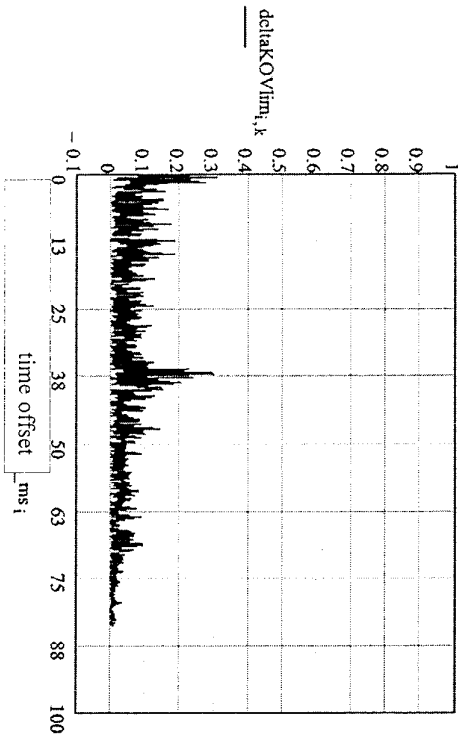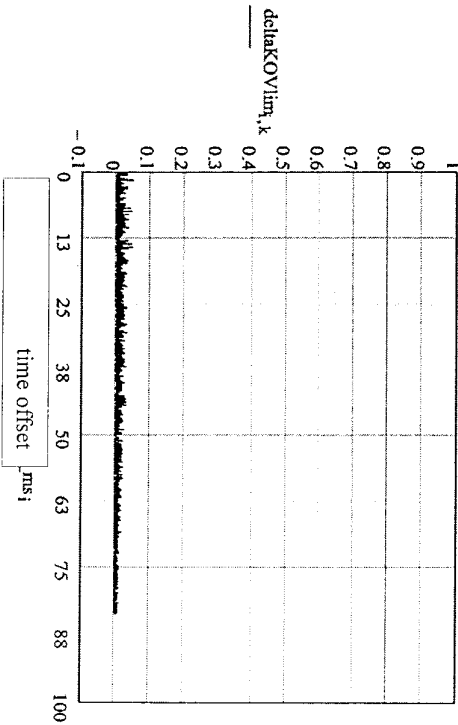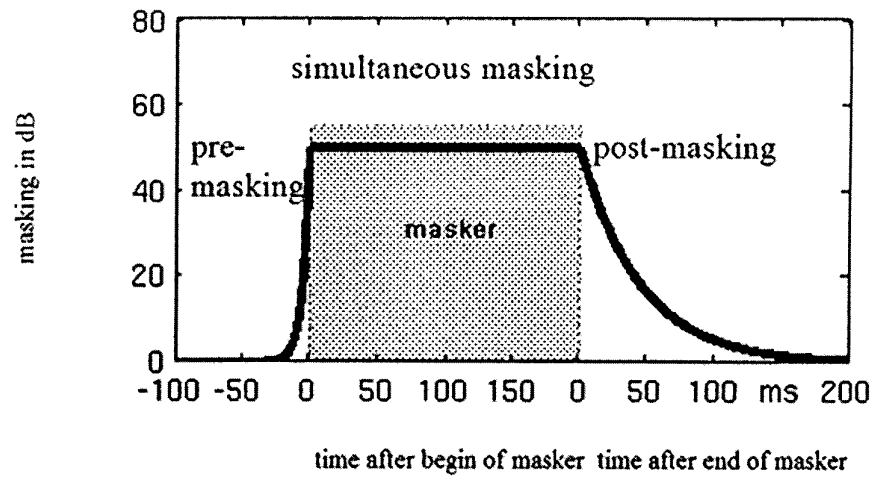
Fig. 11a

Fig. 11b

Fig. 12

Fig. 13

sum signal spectrum $\overline{i,k}$

difference signal spectrum i,k
.....

Fig. 14

sum signal spectrum $_{i,k}$

difference signal spectrum $_{i,k}$

median sum $_{i,k}$

Fig. 15

sum signal spectrum $_{i,k}$

difference signal spectrum $_{i,k}$

Fig. 16

difference signal spectrum $_{i,k}$

median difference $_{i,k}$

LM difference $_{i,k}$

Fig. 17a

sum signal spectrum $_{i,k}$

difference signal spectrum $_{i,k}$
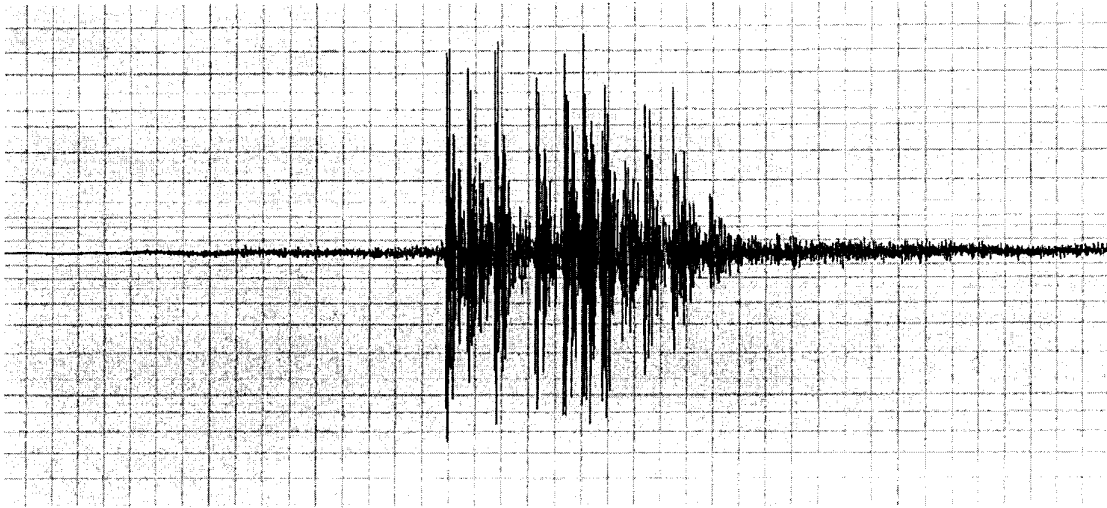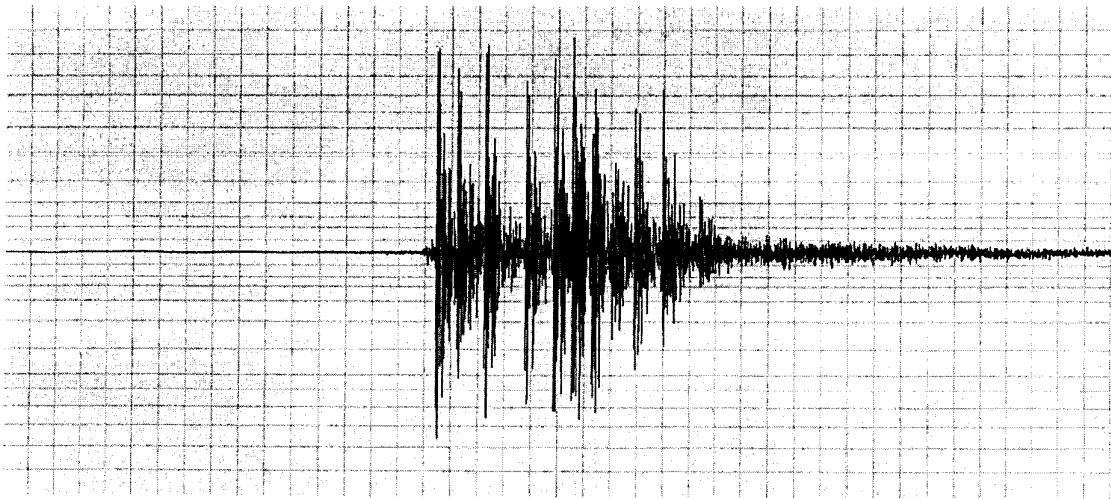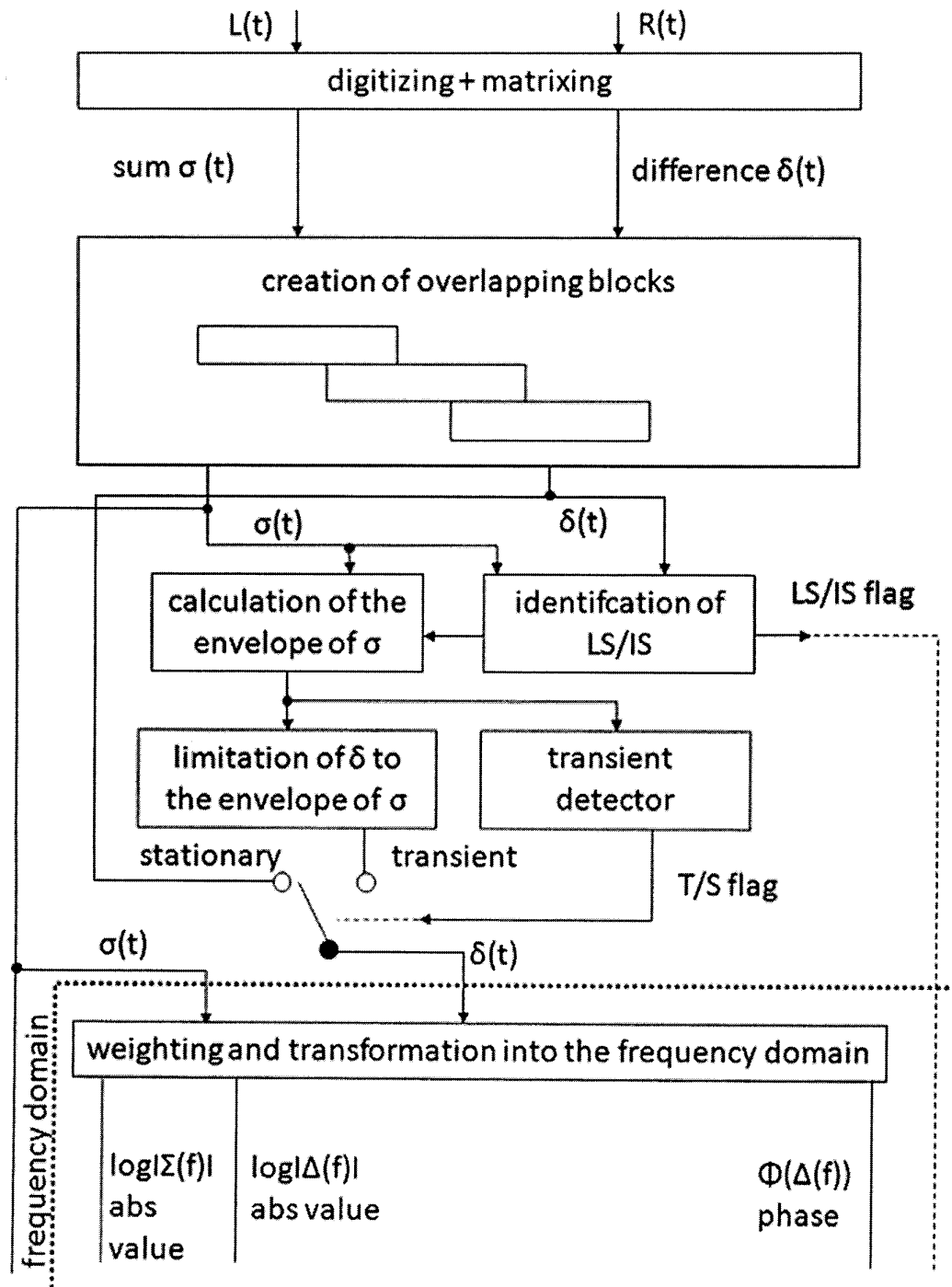
LM sum $_{i,k}$

LM difference $_{i,k}$
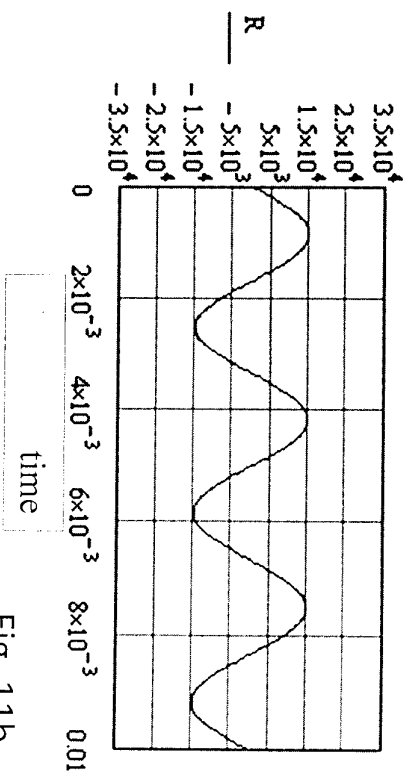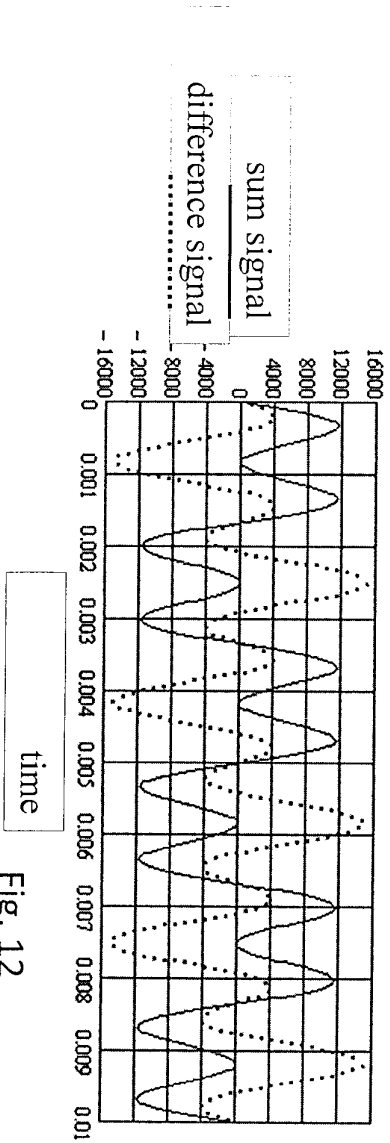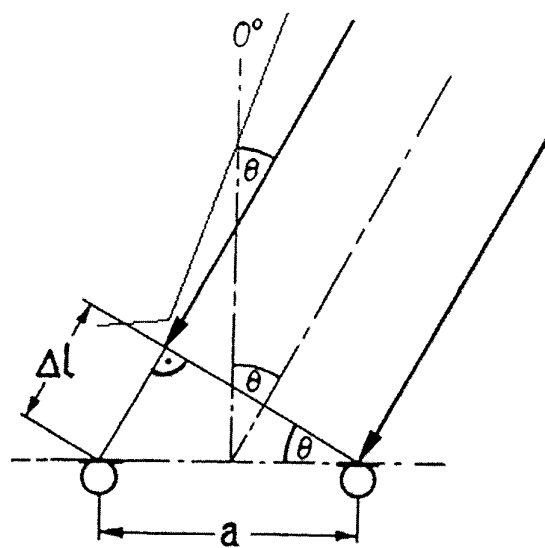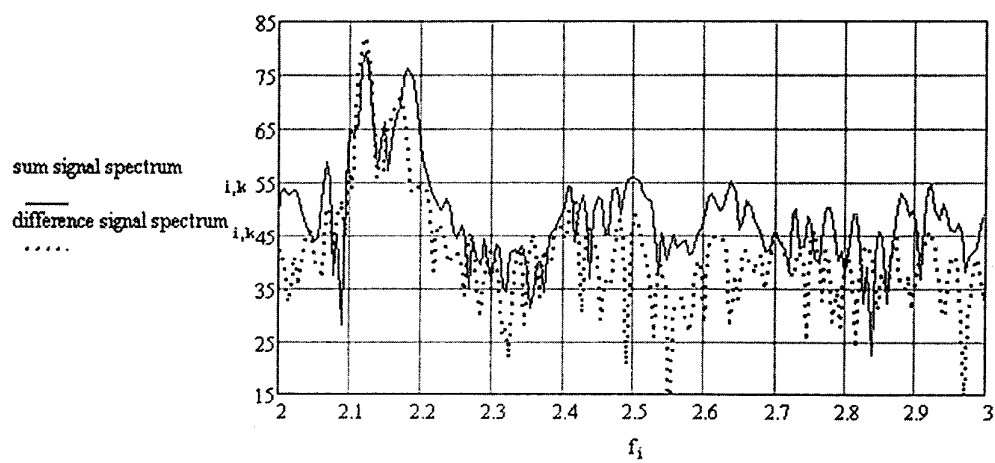
Fig. 17b

Fig. 18a

Fig. 18b

Fig. 19
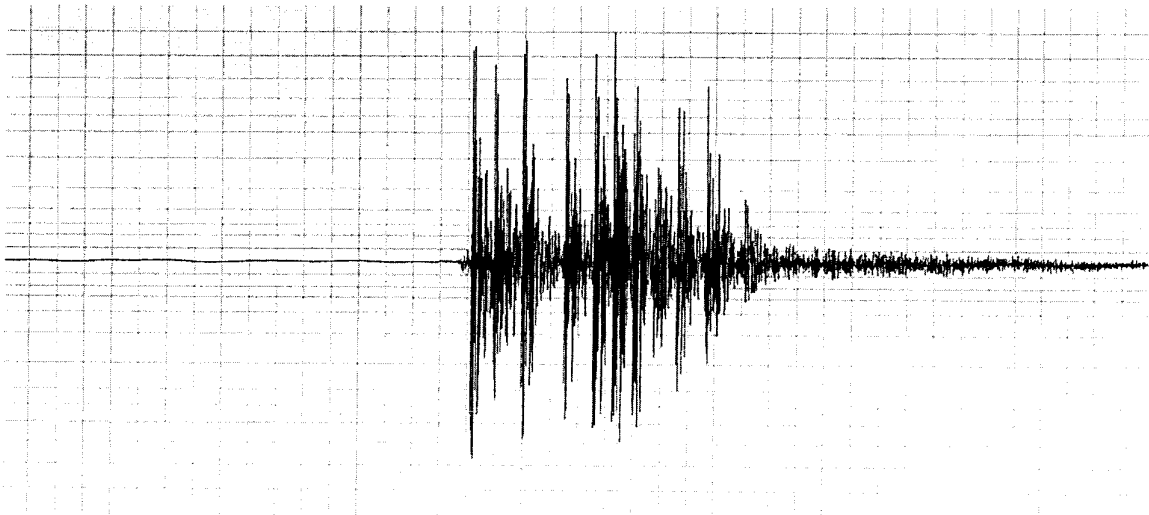
Fig. 20

Fig. 21

Fig. 22

Fig. 23a

Fig. 23b

$\log |\Sigma(f)|$

creation of a reference curve by e.g. median filtering over n_1 values

$\log |\Sigma(f)|_{ref1}$

+

distance 1

distance 1 > threshold 1?

no

yes

f_A

$\neq f_A$

reduction of $\log |\Delta(f)|$ to median according to Fig. 23b

reduction of $\log |\Delta(f)|$ to $\log |\Sigma(f)|$ according to Fig. 23b

Fig. 24

Fig. 25