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Bandwidth Extension Patent Overview

Here we present a chronological overview of BWE-related patents, from 1943 to 2004. This overview has resulted from searches by the authors, and is not assumed to cover *all* BWE patents. Specifically, the overview is limited to US patents; also, the kind of patents included cover those areas that are more or less closely related to the material covered in this book (including both low- and high-frequency BWE methods). As a resource, it is hoped that this patent list will complement the list of cited references (these patents are therefore not separately listed in the bibliography).

Each item in the list presents the key data for the patent (title and US patent number, inventor, assignee, and date[1]), and is believed to be accurate but not guaranteed to be so. The abstract is in most cases directly copied from the published abstract; deleted passages are marked as (. . .). Full text for US patents can be obtained from the US Patent and Trademark Office (USPTO), which also maintains a website with a searchable database (http://www.uspto.gov).

PSEUDO-EXTENSION OF FREQUENCY BANDS

Title: Pseudo-extension of frequency bands (2,315,248).
Inventor: Louis A. de Rosa.
Date: March 30, 1943

This invention deals with the pseudo-extension of frequency bands and particularly with improvements in the method and means wherein an audio signal, at some point or at some time in its transmission either directly or indirectly to the ear, is modified so that, while all the composite frequencies present in the original audio signal are not present in the signal ultimately transmitted to the ear, the auditory perception is of a sound that has substantially all the sonant characteristics of the original audio signal.

[1]Note that before June 8, 1995, patent protection expired 17 years after the patent was granted. After June 8, 1995, patent protection expires 20 years after the filing date.

Audio Bandwidth Extension: Application of Psychoacoustics, Signal Processing and Loudspeaker Design. E. Larsen and R. M. Aarts
 ISBN 0-470-85864-8

APPARATUS FOR IMPROVING SOUNDS OF MUSIC AND SPEECH

Title: Circuit for simulating string bass sound (2,866,849).
Inventor: Charles D. Lindridge.
Assignee: one-fourth to L.C. Krazinski.
Date: December 30, 1958.

This invention relates to sound-reproducing systems, sound-reinforcing systems, music and speech, and specifically to apparatus for improving sounds of music and speech in which there is a deficiency of high frequencies received from the sound source. The improvement is made by producing harmonics of frequencies higher than 3 kc in the sounds and producing sounds at frequencies higher than 6 kc at a loudspeaker.

An object of this invention is to compensate for loss in sound at high frequencies due to the use of a narrowband of frequencies in a transmission system or due in part to the greater directivity of high frequencies in air relative to that at low frequencies, and in part to the greater absorption of sound in air at normal room temperature and humidity at high frequencies than at low frequencies.

(. . .)

AUDIO TRANSMISSION NETWORK

Title: Audio transmission network (2,379,714).
Inventor: R.L. Hollingsworth.
Assignee: Radio Corporation of America, Del.
Date: July 3, 1945

It is an object of this invention to improve the intelligibility or overall quality of a band-limited audio signal, such as that received from radio transmissions.

CIRCUIT FOR USE IN MUSICAL INSTRUMENTS

Title: Circuit for use in musical instruments (3,006,228).
Inventor: J. P. White.
Date: October 31, 1961.

This invention relates to certain novel circuit arrangements that may be used to produce a pleasing musical effect (. . .) to produce tones rich in overtones or quality.

ARTIFICIAL RECONSTRUCTION OF SPEECH

Title: Artificial reconstruction of speech (3,127,476).
Inventor: Edward E. David.
Assignee: Bell Telephone Laboratories, Incorporated, New York, N.Y.
Date: March 31, 1964.

This invention relates to the reconstruction of artificial speech from narrowband transmitted signals, and has for its principal object the improvement of quality of such artificial speech.

TONE GENERATION SYSTEM

Title: Tone generation system (3,213,180).
Inventor: Jack C. Cookerly and George R. Hall.
Date: October 19, 1965.

The object of the invention is 'to provide a novel tone generation system in which output tones are derived from the normal sound of the instrument, but in which such output tones may have an entirely different quality so that the known instrument may be used to generate tones sounding completely different from those characterizing the instrument', but are 'nevertheless characterized by the manner in which the initiating natural tone of the instrument is played by the musician'.

ELECTRICAL WOODWIND MUSICAL INSTRUMENT

Title: Electrical woodwind musical instrument having electronically produced sounds for accompaniment (3,429,976).
Inventor: Daniel J. Tomcik.
Assignee: Electro-Voice Incorporated, Buchanan, Mich.
Date: February 25, 1969.

A monophonic wind-type instrument is disclosed, employing a piezoelectric pickup communicating with the air column of the instrument. The piezoelectric pickup is utilized to generate electrical signals that are amplified, filtered as to tone, and reproduced by a loudspeaker. The electrical signals are also shaped and utilized to drive a pulse generator to produce multiples and sub-multiples of the frequency of the tone produced by the woodwind instrument. The output of the divider is tone filtered to produce a voice independent of the wind instrument. A gate circuit is provided between the trigger circuit and divider to delay in actuation of the divider following initial production of a tone by the instrument in order to avoid spurious mechanically excited electrical outputs.

MUSICAL INSTRUMENT ELECTRONIC TONE PROCESSING SYSTEM

Title: Musical instrument electronic tone processing system (3,535,969).
Inventor: David A. Bunger.
Assignee: D.H. Baldwin Company, Cincinnati, Ohio.
Date: October 27, 1970.

An audio tone from a musical instrument is applied to a phase splitter to provide two oppositely phased tone signals, each of which is passed to a respective field effect transistor transmission gate. In addition, the audio tone is converted to a square wave having the same frequency as the fundamental frequency of the tone. The square wave is passed to a frequency divider, which provides a pair of gating signals for the respective transmission gates, the gating signals being oppositely phased and at half the fundamental frequency of the tone. The oppositely phased tone signals are alternately passed by the gates and combined in a tone colour filter circuit, which imparts specified musical tone qualities to the combined signal. In addition, the original tone may be gated via a further field effect transistor gate by a signal having a frequency that is one-quarter of the tone fundamental frequency, the gated signal being passed to an appropriate tone colour filter. The original tone is also passed directly to a tone colour filter. All of the filtered signals are then amplified and passed to a loudspeaker system to provide an acoustic signal of substantially greater tonal complexity than the original tone and which is controlled in frequency and amplitude by the frequency and amplitude respectively in the input tone.

OCTAVE JUMPER FOR MUSICAL INSTRUMENTS

Title: Octave jumper for musical instruments (3,651,242).

Inventor: Chauncey R. Evans.

Assignee: Columbia Broadcasting System Inc. N.Y.

Date: March 21, 1972.

A bass or other guitar has a transducer for each of its strings, and each transducer is connected to an octave-jumping circuit that lowers the musical tone produced by the individual string, all without loss of either harmonics or amplitude variations. The waveform of the fundamental frequency of each musical tone is squared, divided by two and then amplitude modulated to follow the amplitude envelope of the original tone. The modulated square wave contains only odd harmonics of the lowered frequency fundamental. The missing even harmonics are restored by combining with the modulated square wave the original tone containing all of its harmonics.

SPEECH QUALITY IMPROVING SYSTEM

Title: Speech quality improving system utilizing the generation of higher harmonic components (3,828,133).

Inventors: Hikoichi Ishigami *et al.*

Assignee: Kokusai Denshin Denwa Kabushiki Kaisha, Tokyo-To, Japan.

Date: August 6, 1974.

A speech quality–improving system for a band-limited voice signal, comprising a branching circuit for dividing the band-limited voice signal into two branched signals, each having the same waveform as the band-limited voice signal, a higher harmonic signal generator for generating higher harmonic components of one of the two branched signals, and a combining circuit for combining the other of the two branched signals with the generated higher harmonic components to provide a combined voice signal having an improved speech quality realized by increasing the higher harmonic components of the bandlimited voice signal. The higher harmonic generator comprises a cascade combination of an instantaneous compressor and a level range expander having reciprocal power characteristics of the compression ratio of the instantaneous compressor.

AUDIO SIGNAL PROCESSOR

Title: Audio signal processor (4,144,581).

Inventor: Andrew Prudente.

Date: March 13, 1979.

Random audio signals are converted into sine waves of corresponding frequency and duration, and harmonics are derived from the sine waves. The harmonics are controllably attenuated and selectively inverted and combined to form an output signal. The conversion of the random signals is effected by squaring the same to drive a Schmitt trigger that feeds into a levelled integrator that leads to a diode function generator. The harmonics are generated with the use of four-quadrant multipliers.

CIRCUIT FOR SIMULATING STRING BASS SOUND

Title: Circuit for simulating string bass sound (4,175,465).

Inventor: George F. Schmoll.

Assignee: CBS Inc., New York, N.Y.

Date: November 27, 1979.

In a circuit for simulating the sound produced when a stringed instrument, such as a bass viol, is plucked, square wave signals of different frequencies from a tone generator are combined to produce a synthesized saw-tooth wave form, which is applied to a low-pass filter to remove the extremely high order harmonics, and then applied to an amplifier the gain of which is controlled in accordance with an envelope signal having a fast attack and a relatively slow decay. The resulting amplified signal is applied to an off-centre-biased amplifier that alters the harmonic content of the output signal as a function of decay time such that when the signal is acoustically reproduced it closely simulates the sound produced when a bass viol string is plucked.

DETECTION AND MONITORING DEVICE

Title: Detection and monitoring device (4,182,930).

Inventor: David E. Blackmer.

Assignee: dbx Inc., Newton, MA.

Date: January 8, 1980.

An improved audio signal processing system synthesizes from an audio signal, an enhanced audio signal by sensing signal energy of the audio signal within a preselected frequency portion of the audio signal, dividing the sensed signal energy into a plurality of discrete bands according to the frequency thereof and generating, responsively to the signal energy in each of the bands, a like plurality of second signals each of which includes frequency components that are sub-harmonics of the frequencies of the corresponding frequency band. The second signals are combined so as to provide a combined signal and the latter is added to the audio signal to provide the enhanced audio signal.

FREQUENCY CONVERSION SYSTEM

Title: Frequency conversion system of tone signal produced by electrically picking up mechanical vibrations of musical instruments (4,233,874).

Inventor: Rokurota Mantani.

Assignee: Nippon Gakki Seizo Kabushiki Kaisha, Hamamatsu, Japan.

Date: November 18, 1980.

An octave conversion system of the fundamental frequency of an audible tone signal produced by electrically picking up mechanical vibration of a musical instrument in which the audible tone signal and an audible modulation signal having a frequency in a preselected relation to the fundamental frequency of the tone signal are applied to a multiplier, which is preferably constituted by a voltage-controlled amplifier. When the modulation signal has a frequency half that of the tone signal, the tone signal is one-octave down-converted, while, when the modulation frequency is equal to the tone signal frequency the tone signal is one-octave upconverted. With this frequency conversion system, the fundamental wave component of the octave-converted tone signal has the same envelope as that of the original tone signal. This frequency conversion system is advantageous in attaining small-size versions of electric musical instruments and extension of inherent compasses of electric musical instruments.

AUDIO PROCESSING SYSTEM FOR RESTORING BASS FREQUENCIES

Title: Audio processing system for restoring bass frequencies (4,698,842).

Inventor: Gregory C. Mackie *et al.*

Assignee: Electronic Engineering and Manufacturing, Inc., Lynnwood, Wash.

Date: October 6, 1987.

An audio processing system for injecting left and right channel audio signals with a signal having a fundamental frequency component that is half the frequency of the highest amplitude low-frequency component in the left and right channel audio signals. The left and right channel audio signals are combined to form a monaural signal that is low-pass filtered and applied to a demodulator circuit. The demodulator circuit generates a control signal having a frequency that is half the frequency of the highest amplitude frequency component in the signal at the output of the band-pass filter. The control signal varies the phase of the signal at the output of the band-pass filter according to the polarity of the control signal. The resulting signal is selectively added to the left and right input signals. In order to prevent the audio processing circuit from producing annoying artefact when the audio signals are vocally generated, a voice detector determines that the input signals are from a vocal source and then disables the audio processing circuit. The voice detector operates by comparing the monaural (left plus right) signal to a differential signal (left minus right). Vocal source material has a relatively higher monaural signal, while a musical source has a relatively higher differential signal.

SIGNAL SYNTHESIZER

Title: Signal synthesizer (4,700,390).

Inventor: Kenji Machida.

Date: October 13, 1987.

To enhance low- and high-frequency components in a sound signal, low-frequency components are used to generate new yet lower frequencies (sub-harmonics), and high-frequency components are used to generate new yet higher frequencies (harmonics), the new frequencies added to the original signal thereby increasing the original signal bandwidth.

METHOD TO ELECTRONICALLY CLARIFY SOUND OR PICTURE INFORMATION

Title: Method to electronically clarify sound or picture information and an arrangement to carry out the method (4,731,852).

Inventor: Liljeryd, Lars G.

Date: March 15, 1988

A method for electronically clarifying sound or picture information and an arrangement for carrying out the method. It was previously known to generate harmonics and sub-harmonics of a useful signal within an audio or video frequency band and to add these to the useful signal in order to improve the perceptibility. Undesirable intermodulation products are generated, however, particularly the difference intermodulation products and the non-linear amplitude ratio between generated harmonic components related to the input signal. The suggested method eliminates substantially all of these undesirable intermodulation products completely and provides a linear amplitude ratio by forming two orthogonal components from the useful signal, compressing one or both of these components and multiplying the result to form the harmonics that are thereafter mixed with the useful signal.

LOW-PITCHED SOUND CREATOR

Title: Low-pitched sound creator (4,790,014).

Inventors: Koji Watanabe *et al.*

Assignee: Matsushita Electric Industrial Co., Ltd., Osaka, Japan.

Date: December 6, 1988.

An analog sound signal is outputted. Low-frequency components are selected from the outputted analog sound signal so that a low-pitched sound signal is derived from the analog sound signal. A key of the low-pitched sound signal is lowered so that a very low pitched sound signal is derived from the low-pitched sound signal. The analog sound signal and the very low pitched sound signal may be converted into corresponding sounds respectively.

SUB-HARMONIC TONE GENERATOR FOR BOWED MUSICAL INSTRUMENTS

Title: Sub-harmonic tone generator for bowed musical instruments (4,856,401).

Inventor: Richard E. D. McClish.

Date: August 15, 1989.

A device to produce sub-harmonic tone signals in response to a tone signal from a transducer having preferably maximum sensitivity in the plane of bowing of a bowed musical instrument by passing selected cycles of the transducer signal through signal gates that are controlled jointly by sub-harmonic control signals at sub-multiples of the fundamental frequency of the transducer signal and by a signal indicative of the detection of a fundamental frequency. Each sub-harmonic tone signal thus produced has a tone colour, which approximates that of the corresponding bowed musical instrument of the same frequency range and which is independent of the direction of bowing.

LOW FREQUENCY AUDIO DOUBLING AND MIXING CIRCUIT

Title: Low frequency audio doubling and mixing circuit (EP0546619).

Inventor: Wayne Schott.

Assignee: US Philips Corporation, NY

Priority Date: December 9, 1991.

A circuit for doubling and mixing low-frequency audio signals includes an input for receiving an audio signal having a substantially wide frequency range, a circuit coupled to said input means for separating signal components in a low-frequency band of the audio signal from the wide frequency range thereof, a frequency doubler coupled to the separating circuit for doubling the frequencies of the signal components in the low-frequency band, and a mixer for mixing the frequency-doubled signal components with the input audio signal, whereby the signal components in the low-frequency band now also appear one octave higher.

MUSIC TONE PITCH SHIFT APPARATUS

Title: Music tone pitch shift apparatus (5,131,042).

Inventor: Mikio Oda.

Assignee: Matsushita Electric Industrial Co., Ltd., Osaka, Japan.

Date: July 14, 1992.

A music tone pitch shift apparatus that converts an original audio signal into digital data by way of pulse code modulation (PCM), shifting the pitch, and converting the pitch-shifted digital data into an analog signal. The PCM digital data is stored in a ring memory at a given sampling speed, and is read out of the memory by a pair of identical read circuits at a common read-addressing speed corresponding to the desired pitch. One of the read circuits starts reading from the opposite address location to the other on the ring memory. Since the read-addressing speed is set faster than the write-addressing speed when increasing the pitch, and vice versa, overtaking or lapping between the addresses could occur. In switching the read circuits from a now-outputting side to a switching-to side alternately, the read address on the switching-to side circuit is stopped increasing at an address location where a zero-amplitude data has been read, until a zero-amplitude data in phase with that which the switching-to side circuit has read is read by the now-outputting side circuit and the switching is made, immediately before the overtaking or lapping occurs on the now-outputting side circuit. Thus, a smooth connection of the pitch-shifted audio signals can be made without including such amplitude-modulated components as in the cross fade method, and therefore, a high-quality music tone pitch shift operation can be realized.

STRING INSTRUMENT SOUND ENHANCING METHOD AND APPARATUS

Title: String instrument sound enhancing method and apparatus (5,218,160).

Inventor: Matthias Grob-Da Veiga.

Date: June 8, 1993.

A sound-enhancing apparatus for use with a string instrument has separate tone pickups for picking up the tones of individual strings and circuits for, determining the fundamental tones of these tones, multiplying the frequencies of these fundamental tones by small integers, and/or dividing them by small integers and consequently producing harmonic overtones and/or undertones. The thus-produced harmonic undertones and/or overtones are selected and amplified according to fixed and/or adjustable criteria and finally admixed with the original sound. The electronic apparatus for performing the process can operate in analog or digital manner.

DIGITAL RECONSTRUCTING OF HARMONICS TO EXTEND BAND OF FREQUENCY RESPONSE

Title: Digital reconstructing of harmonics to extend band of frequency response (5,267,095).

Inventors: T. Hasegawa *et al.*

Assignee: Pioneer Electronic Corporation, Tokyo, Japan.

Date: November 30, 1993.

A PCM digital audio signal playback apparatus is provided for extracting from the digital audio signal read out from a recording medium an original signal component ranging lower than $\frac{1}{2}$ of its sampling frequency f_s, producing a harmonic from the original signal component, extracting a harmonic component ranging higher than $\frac{f_s}{2}$ from the harmonic, and adding the harmonic component to the original signal component. Accordingly, a high-frequency-carrying signal, for example, an impulse, is processed without causing ringings in the waveform response.

TRANSIENT DISCRIMINATE HARMONICS GENERATOR

Title: Transient discriminate harmonics generator (5,424,488).

Inventor: Donn Werrbach.

Assignee: Aphex Systems, Ltd., Sun Valley, Ca.

Date: June 13, 1995.

A transient discriminate harmonics generator that receives an audio input signal and produces an output signal containing harmonics of the input signal. The output signal is amplitude shaped as a function of the input signal's time and amplitude envelope. The present invention, the transient discriminate harmonics generator generally comprises a control circuit for determining a control parameter, and a harmonics-generating circuit regulated by the control circuit for producing an output signal containing harmonics of an input signal, where the transient discriminate harmonics generator first generates a relatively high level of harmonics at an initial occurrence of the input signal, then incrementally reduces the level of harmonies generated during a time period determined by the control parameter following the initial occurrence of the input signal, and finally produces a relatively low level of harmonics after the end of the time period.

SPEECH BANDWIDTH EXTENSION METHOD AND APPARATUS

Title: Speech bandwidth extension method and apparatus (5,455,888).

Inventor: Vasu Iyengar *et al.*

Assignee: Northern Telecom Ltd., Montreal, Canada.

Date: October 3, 1995.

A speech bandwidth extension method and apparatus analyses narrowband speech sampled at 8 kHz using LPC (*authora comment*: linear predictive coding) analysis to determine its spectral shape and inverse filtering to extract its excitation signal. The excitation signal is interpolated to a sampling rate of 16 kHz and analysed for pitch control and power level. A white noise–generated wideband signal is then filtered to provide a synthesized wideband excitation signal. The narrowband shape is determined and compared with templates in respective vector quantizer codebooks, to select respective high-band shape and gain. The synthesized wideband excitation signal is then filtered to provide a high-band signal which is, in turn, added to the narrowband signal, interpolated to the 16-kHz sample rate, to produce an artificial wideband signal. The apparatus may be implemented on a digital signal processor chip.

MUSICAL TONE GENERATING APPARATUS EMPLOYING MICRORESONATOR ARRAY

Title: Musical tone generating apparatus employing microresonator array (5,569,871).

Inventors: James A. Wheaton *et al.*

Assignee: Yamaha Corporation, Japan

Date: October 29, 1996

A musical tone–generating apparatus employs an array of microresonant structures to generate the harmonic component signals of a musical tone to be generated. The microresonant structures produce high-frequency signals that are down-converted to audio-frequency range by mixing them with a high-frequency reference signal. The desired tone colour is achieved by modifying the relative amplitudes of the harmonic component signals to produce a desired tone colour. A large number of microresonators are preferably integrated on a single integrated circuit substrate to provide a variable tone–generating system in a relatively compact environment.

HARMONIC TONE GENERATOR FOR LOW LEVEL INPUT AUDIO SIGNALS

Title: Harmonic tone generator for low level input audio signals and small amplitude input audio signals (5,578,948).

Inventor: Soichi Toyama.

Assignee: Pioneer Electronic Corporation, Tokyo, Japan.

Date: November 26, 1996.

A harmonic tone generator produces a harmonics signal even for input audio signals of small amplitude. Conversion of a digitized audio signal in accordance with a predetermined non-linear function is also performed for an audio signal of small amplitude. According to the second aspect of the invention, a level difference between the digital audio signal level in the present sampling time and the audio signal level in the preceding sampling time is detected and the detected level difference is converted to an output value in accordance with a predetermined non-linear function by a non-linear converting circuit. The converted output value is accumulated. According to the third aspect of the invention, the detected level difference is converted to a function conversion output in accordance with a predetermined function by a non-linear converting circuit. A gain of an amplifier to amplify the audio signal in the present sampling time is changed in accordance with the function conversion output.

LOW FREQUENCY AUDIO CONVERSION CIRCUIT

Title: Low frequency audio conversion circuit (5,668,885).
Inventor: Mikio Oda.
Assignee: Matsushita Electric Industrial Co., Ltd., Osaka, Japan.
Date: September 16, 1997.

A low-frequency audio conversion circuit for converting the frequency of low-frequency audio components. An input audio signal includes a low-frequency audio component lower than the frequency a speaker can reproduce. The low-frequency audio component is filtered and extracted by a low-pass filter and full-wave rectified to generate even-numbered harmonics of the low-frequency audio component. Secondary harmonics are extracted from the even-numbered harmonics and added to the input audio signal after being amplified to an appropriate level. When a speaker whose low-frequency sound reproduction characteristics are poor is used, and a low-frequency component lower than the frequency the speaker can reproduce is supplied, the low-frequency audio component is reproduced as secondary harmonics, which fall within the frequency range of the speaker. Thus, the low-frequency audio component is compensated, and a powerful sound is reproduced at a low cost without degrading the sound.

DIGITAL SIGNAL PROCESSOR FOR ADDING HARMONIC CONTENT

Title: Digital signal processor for adding harmonic content to digital audio signal (5,748,747).
Inventor: Dana C. Massie.
Assignee: Creative Technology, Ltd., Singapore.
Date: May 5, 1998 (The term of this patent shall not extend beyond the expiration date of 5,524,074, which was patented on June 4, 1996).

A digital audio signal processor for adding harmonic content to an input audio signal through a non-linear transfer function with discontinuities. The discontinuities are generated by bit shifting each input value by an amount that is dependent on the sign and magnitude of the input value. The amount by which the input value is shifted is roughly inversely related to the magnitude of the logarithm of the input value. The transfer function is fractal and so provides increased harmonic content for all signal amplitudes.

AUDIO CIRCUIT

Title: Audio circuit (5,771,296).
Inventor: Toyoaki Unemura.
Assignee: Matsushita Electric Industrial Co., Ltd., Osaka, Japan.
Date: June 23, 1998.

An audio circuit for use in a television receiver and the like compensates for the capacity shortage of a speaker box or low-frequency characteristic of the speaker to reproduce vivid and voluminous low-frequency sound. L and R signals of the audio signal are mixed, and then an arbitrary low-frequency band component is extracted therefrom by a filter having an arbitrary frequency characteristic, and extracted component is bisected by a distribution means, and only low-frequency band component is added to the original L and R signals to reproduce the audio signal that is voluminous in a low-frequency band. With a low-frequency band that is difficult to be reproduced by a speaker, the harmonic is stressed by full-wave rectification means to stress low-frequency sound feeling, and when a switching circuit is provided, low-frequency sound stressing by an amplifier and low-frequency sound harmonic stressing by full-wave rectification means can be easily switched.

METHOD AND DEVICE FOR PROCESSING SIGNALS

Title: Method and device for processing signals (5,828,755).
Inventor: E.E. Feremans and F. De Smet.
Date: October 27, 1998.

A method is set forth for processing signals, in particular for treating audio signals, characterized in that it mainly consists in the supply of an input signal to be treated; in the isolation of a number of signals from the input signal, which are mainly situated in a predetermined part of the sound range; in the additional generation of higher harmonics based on the isolated signals; and in the formation of an output signal by combining the signal that contains the generated higher harmonics with at least a part of the above-mentioned input signal, this input signal is either treated or not treated before being combined.

AUDIO SIGNAL PROCESSING CIRCUIT FOR CHANGING THE PITCH OF RECORDED SPEECH

Title: Audio signal processing circuit for changing the pitch of recorded speech (5,848,392)
Inventor: Katsuyuki Shudo.
Assignee: Victor Company of Japan, Ltd., Yokohama, Japan.
Date: December 8, 1998.

A memory has storage segments at different addresses respectively. A write address signal represents an address that is periodically updated at a first frequency. Samples of an audio signal are sequentially written into storage segments of the memory at addresses represented by the write address signal, respectively. A read address signal represents an address that is periodically updated at a second frequency lower than the first frequency. Samples of the audio signal are sequentially read out from storage segments of the memory at addresses represented by the read address signal, respectively. After the address represented by the write address signal overtakes the address represented by the read address signal and until the address represented by the read address signal reaches the address represented by the write address signal that occurs when the address represented by the write address signal overtakes the address represented by the read address signal, inhibition is given of writing of samples of the audio signal into storage segments of the memory at addresses different from the address represented by the write address signal that occurs when the address represented by the write address signal overtakes the address represented by the read address signal.

METHOD AND SYSTEM FOR ENHANCING QUALITY OF SOUND SIGNAL

Title: Method and system for enhancing quality of sound signal (5,930,373)

Inventor: Meir Shashoua and Daniel Glotter.

Assignee: K.S. Waves Ltd., Tel Aviv, Israel.

Date: July 27, 1999.

An apparatus for conveying to a listener a pseudo low-frequency psycho-acoustic sensation (Pseudo-LFPS) of a sound signal, including: frequency unit capable of deriving from the sound signal high-frequency signal and low-frequency signal (LF signal) that extends over a low-frequency range of interest. Harmonics generator coupled to the frequency generator and being capable of generating, for each fundamental frequency within the low-frequency range of interest, a residue harmonic signal having a sequence of harmonics. The sequence of harmonics, generated with respect to each fundamental frequency contains a first group of harmonics that includes at least three consecutive harmonics from among a primary set of harmonics of the fundamental frequency. Loudness generator coupled to the harmonics generator and being capable of bringing the loudness of the residue harmonics signal to match the loudness of the low-frequency signal. Summation unit capable of summing the residue harmonic signal and the high-frequency signal so as to obtain psycho-acoustic alternative signal.

ULTRA BASS

Title: Ultra bass (6,134,330).

Inventor: Gerrit F.M. De Poortere *et al.*

Assignee: US Philips Corporation, NY

Date: October 17, 2000

To improve the perceived audio signal, it is known to use a harmonics generator to create the illusion that the perceived audio includes lower-frequency signal parts than really available. In addition to improving the perceived so-called ultra bass signals (for example 20–70 Hz), the signals in the frequency band between the ultra bass signal and the normal audio signal are also improved.

IMPROVING THREE DIMENSIONAL AUDIO POSITIONING

Title: Method for introducing harmonics into an audio stream for improving three dimensional audio positioning (6,215,879)
Inventor: M.J. Dempsey
Date: April 10, 2001

Method for introducing harmonics into an audio stream for improving three-dimensional audio positioning. The method adds high-frequency harmonics into sampled sound signals to replace high-frequency sound components eliminated before sampling. By adding high-frequency harmonics into the sampled sound signals, a 'richer sound' will be produced. The resulting sampled sound signals will have a frequency spectrum containing a larger number of frequencies. Thus, the ear will have more cues to better position the sampled sound signals.

SYSTEM AND METHOD FOR IMPROVING CLARITY OF AUDIO SYSTEMS

Title: System and method for improving clarity of audio systems (6,335,973)
Inventor: Eliot M. Case
Assignee: Qwest Communications International Inc.
Date: January 1, 2002

A system and method for improving the clarity of an audio signal selects frequencies of the audio signal for processing and adds even harmonic distortion to the selected frequencies, preferably, of at least the second order. The system and method are particularly suited for hearing aid, voice messaging, and telephony applications. In addition, the system and method may be applied to other very low bandwidth signals, such as data-compressed audio signals, computer voice files, computer audio files, and numerous other technologies that have an audio response less than normal human perception. The technique also applies to the use of perceptually coded audio.

PSEUDO-EXTENSION OF FREQUENCY BANDS

Title: Pseudo-extension of frequency bands (6,424,939).
Inventor: Jürgen Herre *et al.*
Assignee: Fraunhofer-Gesellschaft
Date: July 23, 2002

A method for coding or decoding an audio signal combines the advantages of TNS processing and noise substitution. A time-discrete audio signal is initially transformed to the frequency domain in order to obtain spectral values of the temporal audio signal. Subsequently, a prediction of the spectral values in relation to frequency is carried out in order to obtain spectral residual values. Within the spectral residual values, areas are detected encompassing spectral residual values with noise properties. The spectral residual values in the noise areas are noise-substituted, whereupon information concerning the noise areas and noise substitution is incorporated into side information pertaining to a coded audio signal. Thus, considerable bit savings in case of transient signals can be achieved.

AUDIO SYSTEM

Title: Audio system (6,678,380).
Inventor: R.M. Aarts
Assignee: US Philips Corporation, NY
Date: January 13, 2004.

An audio system includes a circuit for processing an audio signal, this circuit having an input for receiving the audio signal and an output for supplying an output signal. The circuit further includes a harmonics generator coupled to the input for generating harmonics of the audio signal, and an adding circuit coupled to the input as well as to the harmonics generator for supplying a sum of the audio signal and the generated harmonics to the output. The harmonics generator is embodied so as to limit the amplitude of the generated harmonics.

SOURCE CODING ENHANCEMENT USING SPECTRAL-BAND REPLICATION

Title: Source coding enhancement using spectral-band replication (6,680,972).
Inventor: L.G. Liljeryd *et al.*
Assignee: Coding Technologies Sweden AB
Date: January 20, 2004

The present invention proposes a new method and apparatus for the enhancement of source-coding systems. The invention employs bandwidth reduction prior to or in the encoder, followed by spectral-band replication at the decoder. This is accomplished by the use of new transposition methods, in combination with spectral envelope adjustments. Reduced bit rate at a given perceptual quality or an improved perceptual quality at a given bit rate is offered. The invention is preferably integrated in a hardware or software codec, but can also be implemented as a separate processor in combination with a codec. The invention offers substantial improvements practically independent of codec type and technological progress.

SOUND AND VISION SYSTEM

This is a European patent application, included for reference (see Sec. I.2.1).

Title: Sound and vision system (European patent application EP02708577)
Inventor: R.M. Aarts and M.T. Johnson
Assignee: Royal Philips Electronics N.V.
Date: Filed March 20, 2002

A sound and vision system comprising a display device and an acoustic transducer means, such as a loudspeaker or a microphone. The display device includes display cells having opposite electrodes and includes a conductive means connected to the electrodes in order to address said display cells. The acoustic transducer means is formed by a display cell and the conductive means is electrically coupled to the acoustic transducer means in order to convey signals, as a result of which the acoustic transducer means is an integral part of the display device.