CPC COOPERATIVE PATENT CLASSIFICATION

G PHYSICS
(NOTES omitted)

INSTRUMENTS

G10 MUSICAL INSTRUMENTS; ACOUSTICS
(NOTES omitted)

G10L SPEECH ANALYSIS OR SYNTHESIS; SPEECH RECOGNITION; SPEECH OR VOICE PROCESSING; SPEECH OR AUDIO CODING OR DECODING

NOTE

This subclass does not cover:
- devices for the storage of speech signals, which are covered by subclasses G11B and G11C;
- encoding of compressed speech signals for transmission or storage, which is covered by group H03M 7/30.

13/00		**Speech synthesis; Text to speech systems**
13/02	.	Methods for producing synthetic speech; Speech synthesisers
2013/021	. .	{Overlap-add techniques}
13/027	. .	Concept to speech synthesisers; Generation of natural phrases from machine-based concepts (generation of parameters for speech synthesis out of text G10L 13/08)
13/033	. .	Voice editing, e.g. manipulating the voice of the synthesiser
13/0335	. . .	{Pitch control}
13/04	. .	Details of speech synthesis systems, e.g. synthesiser structure or memory management
13/043	. . .	{Synthesisers specially adapted to particular applications}
13/047	. . .	Architecture of speech synthesisers
13/06	.	Elementary speech units used in speech synthesisers; Concatenation rules
13/07	. .	Concatenation rules
13/08	.	Text analysis or generation of parameters for speech synthesis out of text, e.g. grapheme to phoneme translation, prosody generation or stress or intonation determination
2013/083	. .	{Special characters, e.g. punctuation marks}
13/086	. .	{Detection of language}
13/10	. .	Prosody rules derived from text; Stress or intonation
2013/105	. . .	{Duration}
15/00		**Speech recognition** (G10L 17/00 takes precedence)
15/005	.	{Language recognition}
15/01	.	Assessment or evaluation of speech recognition systems
15/02	.	Feature extraction for speech recognition; Selection of recognition unit
2015/022	. .	{Demisyllables, biphones or triphones being the recognition units}
2015/025	. .	{Phonemes, fenemes or fenones being the recognition units}
2015/027	. .	{Syllables being the recognition units}
15/04	.	Segmentation; Word boundary detection
15/05	. .	Word boundary detection
15/06	.	Creation of reference templates; Training of speech recognition systems, e.g. adaptation to the characteristics of the speaker's voice (G10L 15/14 takes precedence)
15/063	. .	{Training}
2015/0631	. . .	{Creating reference templates; Clustering}
2015/0633	. . . .	{using lexical or orthographic knowledge sources}
2015/0635	. . .	{updating or merging of old and new templates; Mean values; Weighting}
2015/0636	. . . .	{Threshold criteria for the updating}
2015/0638	. . .	{Interactive procedures}
15/065	. .	Adaptation
15/07	. . .	to the speaker
15/075	. . . .	{supervised, i.e. under machine guidance}
15/08	.	Speech classification or search
2015/081	. .	{Search algorithms, e.g. Baum-Welch or Viterbi}
15/083	. .	{Recognition networks (G10L 15/142, G10L 15/16 take precedence)}
2015/085	. .	{Methods for reducing search complexity, pruning}
2015/086	. .	{Recognition of spelled words}
2015/088	. .	{Word spotting}
15/10	. .	using distance or distortion measures between unknown speech and reference templates
15/12	. .	using dynamic programming techniques, e.g. dynamic time warping [DTW]
15/14	. .	using statistical models, e.g. Hidden Markov Models [HMMs] (G10L 15/18 takes precedence)
15/142	. . .	{Hidden Markov Models [HMMs]}
15/144	. . . .	{Training of HMMs}
15/146	. . . . .	{with insufficient amount of training data, e.g. state sharing, tying, deleted interpolation}
15/148	. . . .	{Duration modelling in HMMs, e.g. semi HMM, segmental models or transition probabilities}
15/16	. .	using artificial neural networks
15/18	. .	using natural language modelling
15/1807	. . .	{using prosody or stress}

15/1815 . . . {Semantic context, e.g. disambiguation of the recognition hypotheses based on word meaning}
15/1822 . . . {Parsing for meaning understanding}
15/183 . . . using context dependencies, e.g. language models
15/187 . . . . Phonemic context, e.g. pronunciation rules, phonotactical constraints or phoneme n-grams
15/19 . . . . Grammatical context, e.g. disambiguation of the recognition hypotheses based on word sequence rules
15/193 . . . . . Formal grammars, e.g. finite state automata, context free grammars or word networks
15/197 . . . . . Probabilistic grammars, e.g. word n-grams
15/20 . Speech recognition techniques specially adapted for robustness in adverse environments, e.g. in noise, of stress induced speech (G10L 21/02 takes precedence)
15/22 . Procedures used during a speech recognition process, e.g. man-machine dialogue
2015/221 . . {Announcement of recognition results}
15/222 . . {Barge in, i.e. overridable guidance for interrupting prompts}
2015/223 . . {Execution procedure of a spoken command}
2015/225 . . {Feedback of the input speech}
2015/226 . . {Taking into account non-speech caracteristics}
2015/227 . . . {of the speaker; Human-factor methodology}
2015/228 . . . {of application context}
15/24 . Speech recognition using non-acoustical features
15/25 . . using position of the lips, movement of the lips or face analysis
15/26 . Speech to text systems (G10L 15/08 takes precedence)
15/265 . . {Speech recognisers specially adapted for particular applications (devices for signalling identity of wanted subscriber in a telephonic communication equipment controlled by voice recognition H04M 1/271; speech interaction details in interactive information services in a telephonic communication system H04M 3/4936)}
15/28 . Constructional details of speech recognition systems
15/285 . . {Memory allocation or algorithm optimisation to reduce hardware requirements}
15/30 . . Distributed recognition, e.g. in client-server systems, for mobile phones or network applications
15/32 . . Multiple recognisers used in sequence or in parallel; Score combination systems therefor, e.g. voting systems
15/34 . . Adaptation of a single recogniser for parallel processing, e.g. by use of multiple processors or cloud computing

17/00 Speaker identification or verification
17/005 . {Speaker recognisers specially adapted for particular applications (G07C 9/00071 takes precedence)}
17/02 . Preprocessing operations, e.g. segment selection; Pattern representation or modelling, e.g. based on linear discriminant analysis [LDA] or principal components; Feature selection or extraction
17/04 . Training, enrolment or model building
17/06 . Decision making techniques; Pattern matching strategies
17/08 . . Use of distortion metrics or a particular distance between probe pattern and reference templates
17/10 . . Multimodal systems, i.e. based on the integration of multiple recognition engines or fusion of expert systems
17/12 . . Score normalisation
17/14 . . Use of phonemic categorisation or speech recognition prior to speaker recognition or verification
17/16 . Hidden Markov models [HMMs]
17/18 . Artificial neural networks; Connectionist approaches
17/20 . Pattern transformations or operations aimed at increasing system robustness, e.g. against channel noise or different working conditions
17/22 . Interactive procedures; Man-machine interfaces
17/24 . . the user being prompted to utter a password or a predefined phrase
17/26 . Recognition of special voice characteristics, e.g. for use in lie detectors; Recognition of animal voices

19/00 Speech or audio signals analysis-synthesis techniques for redundancy reduction, e.g. in vocoders; Coding or decoding of speech or audio signals, using source filter models or psychoacoustic analysis (in musical instruments G10H)
2019/0001 . {Codebooks}
2019/0002 . . {Codebook adaptations}
2019/0003 . . {Backward prediction of gain}
2019/0004 . . {Design or structure of the codebook}
2019/0005 . . . {Multi-stage vector quantisation}
2019/0006 . . . {Tree or treillis structures; Delayed decisions}
2019/0007 . . {Codebook element generation}
2019/0008 . . . {Algebraic codebooks}
2019/0009 . . . {Orthogonal codebooks}
2019/001 . . . {Interpolation of codebook vectors}
2019/0011 . . {Long term prediction filters, i.e. pitch estimation}
2019/0012 . . {Smoothing of parameters of the decoder interpolation}
2019/0013 . . {Codebook search algorithms}
2019/0014 . . . {Selection criteria for distances}
2019/0015 . . . {Viterbi algorithms}
2019/0016 . . {Codebook for LPC parameters}
19/0017 . {Lossless audio signal coding; Perfect reconstruction of coded audio signal by transmission of coding error (G10L 19/24 takes precedence)}
19/0018 . {Speech coding using phonetic or linguistical decoding of the source; Reconstruction using text-to-speech synthesis}
19/0019 . {Vocoders specially adapted for particular applications}
19/002 . Dynamic bit allocation (for perceptual audio coders G10L 19/032)
19/005 . Correction of errors induced by the transmission channel, if related to the coding algorithm

19/008	.	Multichannel audio signal coding or decoding, i.e. using interchannel correlation to reduce redundancies, e.g. joint-stereo, intensity-coding, matrixing (arrangements for reproducing spatial sound H04R 5/00; stereophonic systems, e.g. spatial sound capture or matrixing of audio signals in the decoded state H04S)
19/012	.	Comfort noise or silence coding
19/018	.	Audio watermarking, i.e. embedding inaudible data in the audio signal
19/02	.	using spectral analysis, e.g. transform vocoders or subband vocoders
19/0204	. .	{using subband decomposition}
19/0208	. . .	{Subband vocoders}
19/0212	. .	{using orthogonal transformation}
19/0216	. . .	{using wavelet decomposition}
19/022	. .	Blocking, i.e. grouping of samples in time; Choice of analysis windows; Overlap factoring
19/025	. . .	Detection of transients or attacks for time/frequency resolution switching
19/028	. .	Noise substitution, i.e. substituting non-tonal spectral components by noisy source (comfort noise for discontinuous speech transmission G10L 19/012)
19/03	. .	Spectral prediction for preventing pre-echo; Temporary noise shaping [TNS], e.g. in MPEG2 or MPEG4
19/032	. .	Quantisation or dequantisation of spectral components
19/035	. . .	Scalar quantisation
19/038	. . .	Vector quantisation, e.g. TwinVQ audio
19/04	.	using predictive techniques
19/06	. .	Determination or coding of the spectral characteristics, e.g. of the short-term prediction coefficients
19/07	. . .	Line spectrum pair [LSP] vocoders
19/08	. .	Determination or coding of the excitation function; Determination or coding of the long-term prediction parameters
19/083	. . .	the excitation function being an excitation gain (G10L 25/90 takes precedence)
19/087	. . .	using mixed excitation models, e.g. MELP, MBE, split band LPC or HVXC
19/09	. . .	Long term prediction, i.e. removing periodical redundancies, e.g. by using adaptive codebook or pitch predictor
19/093	. . .	using sinusoidal excitation models
19/097	. . .	using prototype waveform decomposition or prototype waveform interpolative [PWI] coders
19/10	. . .	the excitation function being a multipulse excitation
19/107	. . . .	Sparse pulse excitation, e.g. by using algebraic codebook
19/113	. . . .	Regular pulse excitation
19/12	. . .	the excitation function being a code excitation, e.g. in code excited linear prediction [CELP] vocoders
19/125	. . . .	Pitch excitation, e.g. pitch synchronous innovation CELP [PSI-CELP]
19/13	. . . .	Residual excited linear prediction [RELP]
19/135	. . . .	Vector sum excited linear prediction [VSELP]
19/16	. .	Vocoder architecture
19/167	. . .	{Audio streaming, i.e. formatting and decoding of an encoded audio signal representation into a data stream for transmission or storage purposes}
19/173	. . .	{Transcoding, i.e. converting between two coded representations avoiding cascaded coding-decoding}
19/18	. . .	Vocoders using multiple modes
19/20	. . . .	using sound class specific coding, hybrid encoders or object based coding
19/22	. . . .	Mode decision, i.e. based on audio signal content versus external parameters
19/24	. . . .	Variable rate codecs, e.g. for generating different qualities using a scalable representation such as hierarchical encoding or layered encoding
19/26	. .	Pre-filtering or post-filtering
19/265	. . .	{Pre-filtering, e.g. high frequency emphasis prior to encoding}
21/00		**Processing of the speech or voice signal to produce another audible or non-audible signal, e.g. visual or tactile, in order to modify its quality or its intelligibility** (G10L 19/00 takes precedence)
21/003	.	Changing voice quality, e.g. pitch or formants
21/007	. .	characterised by the process used
21/01	. . .	Correction of time axis
21/013	. . .	Adapting to target pitch
2021/0135	. . . .	{Voice conversion or morphing}
21/02	.	Speech enhancement, e.g. noise reduction or echo cancellation (reducing echo effects in line transmission systems H04B 3/20; echo suppression in hands-free telephones H04M 9/08)
21/0202	. .	{Applications}
21/0205	. . .	{Enhancement of intelligibility of clean or coded speech}
21/0208	. .	Noise filtering
2021/02082	. . .	{the noise being echo, reverberation of the speech}
2021/02085	. . .	{Periodic noise}
2021/02087	. . .	{the noise being separate speech, e.g. cocktail party}
21/0216	. . .	characterised by the method used for estimating noise
2021/02161	. . . .	{Number of inputs available containing the signal or the noise to be suppressed}
2021/02163	. . . . .	{Only one microphone}
2021/02165	. . . . .	{Two microphones, one receiving mainly the noise signal and the other one mainly the speech signal}
2021/02166	. . . . .	{Microphone arrays; Beamforming}
2021/02168	. . . .	{the estimation exclusively taking place during speech pauses}
21/0224	. . . .	Processing in the time domain
21/0232	. . . .	Processing in the frequency domain
21/0264	. . .	characterised by the type of parameter measurement, e.g. correlation techniques, zero crossing techniques or predictive techniques
21/0272	. .	Voice signal separating
21/028	. . .	using properties of sound source
21/0308	. . .	characterised by the type of parameter measurement, e.g. correlation techniques, zero crossing techniques or predictive techniques
21/0316	. .	by changing the amplitude

21/0324	. . .	Details of processing therefor
21/0332	. . . .	involving modification of waveforms
21/034	. . . .	Automatic adjustment
21/0356	. . .	for synchronising with other signals, e.g. video signals
21/0364	. . .	for improving intelligibility
2021/03643	. . . .	{Diver speech}
2021/03646	. . . .	{Stress or Lombard effect}
21/038	. .	using band spreading techniques
21/0388	. . .	Details of processing therefor
21/04	.	Time compression or expansion
21/043	. .	by changing speed
21/045	. . .	using thinning out or insertion of a waveform
21/047	. . . .	characterised by the type of waveform to be thinned out or inserted
21/049	. . . .	characterised by the interconnection of waveforms
21/055	. .	for synchronising with other signals, e.g. video signals
21/057	. .	for improving intelligibility
2021/0575	. . .	{Aids for the handicapped in speaking}
21/06	.	Transformation of speech into a non-audible representation, e.g. speech visualisation or speech processing for tactile aids (G10L 15/26 takes precedence)
2021/065	. .	{Aids for the handicapped in understanding}
21/10	. .	Transforming into visible information
2021/105	. . .	{Synthesis of the lips movements from speech, e.g. for talking heads}
21/12	. . .	by displaying time domain information
21/14	. . .	by displaying frequency domain information
21/16	. .	Transforming into a non-visible representation (devices or methods enabling ear patients to replace direct auditory perception by another kind of perception A61F 11/04)
21/18	. .	Details of the transformation process
25/00		**Speech or voice analysis techniques not restricted to a single one of groups G10L 15/00-G10L 21/00**
25/03	.	characterised by the type of extracted parameters
25/06	. .	the extracted parameters being correlation coefficients
25/09	. .	the extracted parameters being zero crossing rates
25/12	. .	the extracted parameters being prediction coefficients
25/15	. .	the extracted parameters being formant information
25/18	. .	the extracted parameters being spectral information of each sub-band
25/21	. .	the extracted parameters being power information
25/24	. .	the extracted parameters being the cepstrum
25/27	.	characterised by the analysis technique
25/30	. .	using neural networks
25/33	. .	using fuzzy logic
25/36	. .	using chaos theory
25/39	. .	using genetic algorithms
25/45	.	characterised by the type of analysis window
25/48	.	specially adapted for particular use
25/51	. .	for comparison or discrimination
25/54	. . .	for retrieval
25/57	. . .	for processing of video signals
25/60	. . .	for measuring the quality of voice signals
25/63	. . .	for estimating an emotional state
25/66	. . .	for extracting parameters related to health condition (detecting or measuring for diagnostic purposes A61B 5/00)
25/69	. .	for evaluating synthetic or decoded voice signals
25/72	. .	for transmitting results of analysis
25/75	.	for modelling vocal tract parameters
25/78	.	Detection of presence or absence of voice signals (switching of direction of transmission by voice frequency in two-way loud-speaking telephone systems H04M 9/10)
2025/783	. .	{based on threshold decision}
2025/786	. . .	{Adaptive threshold}
25/81	. .	for discriminating voice from music
25/84	. .	for discriminating voice from noise
25/87	. .	Detection of discrete points within a voice signal
25/90	.	Pitch determination of speech signals
2025/903	. .	{using a laryngograph}
2025/906	. .	{Pitch tracking}
25/93	.	Discriminating between voiced and unvoiced parts of speech signals (G10L 25/90 takes precedence)
2025/932	. .	{Decision in previous or following frames}
2025/935	. .	{Mixed voiced class; Transitions}
2025/937	. .	{Signal energy in various frequency bands}
99/00		**Subject matter not provided for in other groups of this subclass**