DIGITAL EMULATION OF ANALOG COMPANDING ALGORITHMS FOR FM RADIO TRANSMISSION

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ABSTRACT

Analog compander systems have been used to suppress the perception of noise in low dynamic range analog signal storage (tape recording) and signal transmission (FM radio). Commercial compander systems have been analyzed with respect to their signal processing requirements. The general structures of single- and multi-band compander systems have been implemented on a high performance audio PC workstation. Audio tests and measurements with the optimized compander algorithms and parameters show very good performance. Even for transmission channels with very low signal-to-noise ratio (SNR of only 40 dB) an optimized digital multi-band compander emulation removes the channel noise perceptively from the output signal of the transmission system.

1. INTRODUCTION

Analog tape recorders as well as analog FM radio transmission systems show an audio dynamic range of only 50 to 70 dB depending on tape material or RF reception. This reduced dynamic range results in a clearly audible noise floor which is very distracting. In order to reduce the noise perception audio compressors have been used prior to the recording or transmitting process. The compressor reduces the dynamic range of the input signal (e.g. 100 dB to 50 dB at a compression ratio of 2) and as a result all signal amplitudes are above the noise threshold of the tape or the FM transmission. During play-back or in the FM receiver an appropriate expander restores the original dynamic range of the signal by applying an attenuation depending on the signal amplitude. This results in an expansion (e.g. from 50 dB to the original 100 dB) of the dynamic range which on the other hand reduces the noise level by 50 dB. With a compression ratio of 2 for example the perceived signal-to-noise ratio (SNR) can be increased by a factor of 2 (e.g. from 50 dB to 100 dB).

This processing is called companding (compression and expansion). Companding is a time-variable processing and thus can cause audible alterations of the processed signal. By using complementary compressor and expander circuitry the original signal indeed can be restored at the output without any alteration if no noise is added in the compressed path. In real companded signals with noise added we find signal alterations and disturbance like noise pumping and breathing as well as distortion caused by dynamic mistracking of the level of the respective compressor and expander control path inputs. Most of these artifacts cannot be perceived because of being psychoacoustically masked by the preceding sounds (below masking threshold).

Several commercial analog compander systems were on the market. Each system was optimized for its main application. Optimization could be done by choosing appropriate frequency processing structures (broadband, sliding band or multi-band) as well as optimizing the time dependent parameters like attack and decay times of the envelope (level) estimator and the compression factors. Well known compander systems for professional applications like studio tape recorders and movie sound recording are the Dolby A and SR type and the TELEFUNKEN Telcom C4 compander as well as the dbx compander. Principles of those companders have be simplified for usage in consumer variants like Dolby B and C, the TELEFUNKEN HighCOM system and the Burwen Noise Eliminator for the use in cassette tape decks.

Studio tape recorders and tape decks are not produced any more and their usage is very limited nowadays to those locations where a considerable amount of tape material is available (e.g. the archives of radio stations). Nevertheless analog compander systems can still be found in wireless microphone systems. All large brands use analog broadband or multi-band companders in their wireless transmitters and complementary expanders in the receivers (e.g. the Sennheiser HiDyn Plus compander or the dbx-like compander circuitry in the Shure wireless FM systems).

As digital signal processing circuitry becomes cheaper but more powerful, smaller, and less power consumptive, many analog circuits are emulated on digital systems. This holds also for compander systems. Most of the above mentioned compander systems are emulated digitally and are used where the original equipment is not available any more (e.g. in the radio-station to restore old analog tape material onto digital media.) The combination of analog FM transmission and digital companding algorithms is also used in the field of wireless microphones. A digital emulation of the analog compander principles yields several advantages:

a) more complex compander algorithms can be realized on a smaller circuit board size

b) compander parameters can be programmed via presets

c) more complex / adaptive signal processing can be used.

All signal processing techniques aim at the reduction of companding artifacts, at the optimal adaptation of the compander settings to the FM transmission characteristics and at increased reliability and convenience of the wireless FM link.

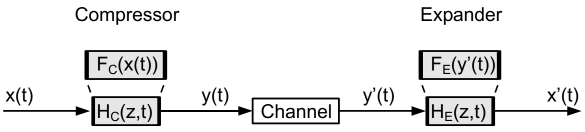

Figure 1: *Compander system.*

2. THEORY OF OPERATION

Compander systems are based on a compressor system before the transmission and expander system at the receiver (see Figure 1). Existing compander systems are built complementary and have time-variant transfer functions $H_C(z,t)$ and $H_E(z,t)$, where the expander transfer function $H_E(z,t)$ is the inverse of the compressor transfer function $H_C(z,t)$ according to

$$H_E(z,t) = \frac{1}{H_C(z,t)}. \tag{1}$$

The notation $H(z,t)$ denotes a time-variant Z-transfer function, because we are aiming at discrete-time realizations for the compressor and the expander. If the compander processing does not meet equation (1), the original signal can not be restored and the compander processing becomes audible. For compander systems the compression factor k is calculated from the signal input level $P_x(t)$ in dB and the signal output level $P_y(t)$ in dB according to

$$k = \frac{P_x(t)}{P_y(t)}. \tag{2}$$

For the noiseless transmission case the compander relation can be written as

$$\begin{aligned} y(t) &= F_C(x(t)) && (3)\\ F_E(y(t)) &= x(t), && (4) \end{aligned}$$

where $F_E(x(t))$ and $F_C(x(t))$ denote the expander and compressor law (gain functions) that depend on the input signal $-1 \leq x(t) \leq 1$. For the compressor law the following relations hold

$$\begin{aligned} F_C(-x(t)) &= -F_C(x(t))\\ F_C(\pm 1) &= \pm 1\\ |F_C(x(t))| &\geq |x(t)|. \end{aligned}$$

Broadband (syllable) companders as well as multi-band (formant) companders use the signal envelope $\text{Env}_x(t)$ instead of the actual signal $x(t)$ to calculate the gain factors [1]. The compressor law of a syllable or formant compander working with the envelope or an appropriate estimate is then defined by

$$\text{Env}_y(t) = F_C(\text{Env}_x(t)) \tag{5}$$

and for the expander law respectively

$$\text{Env}_{x'}(t) = F_E(\text{Env}_{y'}(t)). \tag{6}$$

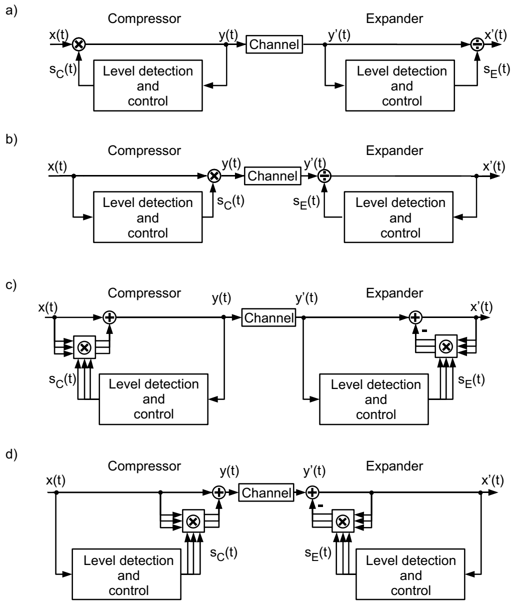

Figure 2: *Feed-back and feed-forward control structures of syllable and formant companders. Single-band multiplicative structure with control signal derived a) from the compressed signal and b) from the original signal. Multi-band additive structure with control signal derived from c) the compressed signal and d) from the original signal.*

The compressed signal $y(t)$ can thus be calculated by the envelope $\text{Env}_x(t)$ of the uncompressed signal or the envelope $\text{Env}_y(t)$ of the compressed signal according to

$$\begin{aligned} y(t) &= \frac{F_C(\text{Env}_x(t))}{\text{Env}_x(t)} \cdot x(t) && (7)\\ y(t) &= \frac{\text{Env}_y(t)}{F_E(\text{Env}_y(t))} \cdot x(t). && (8) \end{aligned}$$

During reconstruction the expanded signal $x'(t)$ is calculated accordingly from the received compressed signal $y'(t)$ with the envelope of the input or the output of the expander given by

$$\begin{aligned} x'(t) &= \frac{F_E(\text{Env}_{y'}(t))}{\text{Env}_{y'}(t)} \cdot y'(t) && (9)\\ x'(t) &= \frac{\text{Env}_{x'}(t)}{F_C(\text{Env}_{x'}(t))} \cdot y'(t). && (10) \end{aligned}$$

These different approaches result in different possible processing structures for syllable and formant companders which are shown in Figure 2. The control signals $s_C(t)$ and $s_E(t)$ are the so-called gain factors derived from the signal envelopes by nonlinear mappings based on the compression factor k.

Those structures that derive the control signal from the compressed signal are favoured in real applications because the envelope detector operates on a signal with reduced dynamic range, especially when the level detector contains true RMS (root mean square) processing. Working on signals with reduced dynamic range imposes less demands on the dynamic range of the analog circuitry or the digital word length [2]. According to [1] both structures yield equal results. The input/output characteristics of a compressor or expander are shown in Figure 3. The upper curve pair

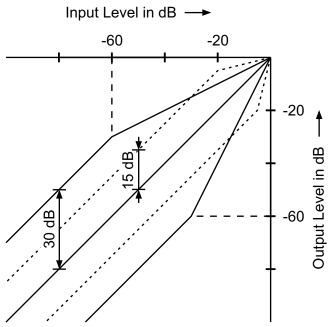

Figure 3: *Example compressor and expander laws. Solid line: compression threshold -60 dB, maximum gain 30 dB, compression ratio 2, compression onset 0 dB. Dotted line: compression threshold -20 dB, maximum gain 15 dB, compression ratio 4, compression onset 0 dB.*

describes the compressor law and the lower two curves describe the expander law. The compressor input and the expander output are drawn in dB on the abscissa. The compressor output and the expander input are drawn in dB on the ordinate.

2.1. Envelope and level estimators

In real compander systems the envelope is approximated by functions like

1. the RMS of the signal amplitude with integration time constant
2. a peak amplitude detection with attack and decay time constants
3. a modified peak detection with multiple decay time constants

All approximations comprise a temporal behaviour caused by the integration (by low-pass filtering) of the squared or absolute value input signal. The deviations also differ according to the frequency content of the input signal. Figure 4 gives an impression of the envelope estimates of the various kinds for two sine burst signals with different frequencies.

2.2. Problems with existing companders

There are several unwanted side effects which are more dominant in single-band companders compared to multi-band companders. The time variable processing leads to distortions in the compressed signal that sometimes cannot be recovered by the expander. This behavior is called dynamic mistracking and leads to more perceived roughness of the low frequency content. This is caused by modulation with the envelope of the high frequency content and vice versa. The temporal integration of the control factor on the other hand leads to pumping and breathing sounds perceived after a sudden mute of a loud signal. The integration/decay time constant has to be large to not follow a low frequency amplitude. On the other hand it has to be as fast as the decay of the post masking curve so that a sudden silence causes the expander to attenuate the output signal resulting in the system noise floor always being below threshold.

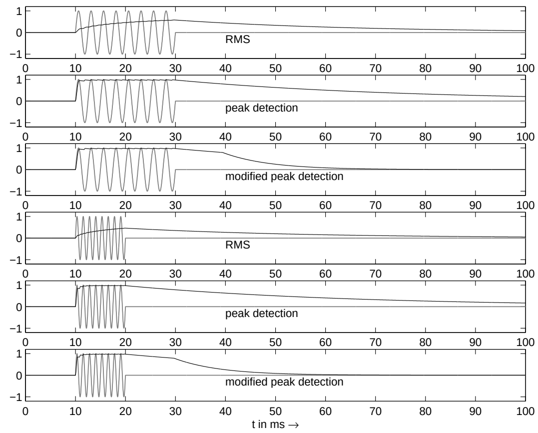

Figure 4: *Temporal behavior of different envelope estimates with attack and release time constants.*

3. EXAMPLES OF ANALOG COMPANDERS

Only the most important existing compander systems from Dolby, TELEFUNKEN and dbx are explained in detail here. Less known companders are the Toshiba Address-System and the JVC Super-ANRS as well as the TELEFUNKEN derivate HighCOM II.

3.1. Dolby A and Dolby SR

The block diagram of the Dolby A compander is shown in Figure 5. It is a four-band system with cut-off frequencies at 80 Hz and 3000 Hz and two high-pass filters working in parallel with cut-off frequencies of 3 kHz and 9 kHz. The compression curves are shown in Figure 6.

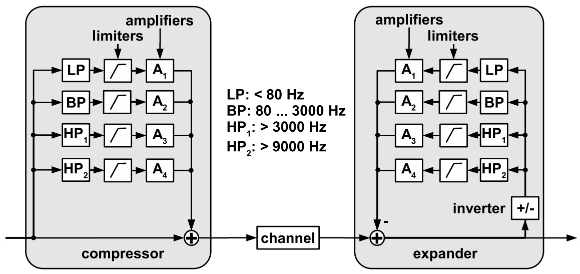

Figure 5: *Block diagram of the Doby A compander.*

This compressor was used for tape recording and could not be used for FM radio because of its non-linear characteristics that resulted in frequency response and signal distortion if the expander received a compressed signal level differing from the compressor output level. Compressor and expander input levels had to be adjusted carefully with an adjustment signal before operation. Usually compressor and expander worked with identical circuitry that could be switched to operate on input (compressor) or on output (expander).

A more sophisticated compander is the Dolby SR (Spectral Recording) system that was used for high end studio tape recording. A block diagram is shown in Figure 7. It is working with fixed and variable filter cut-off frequencies at three different activating

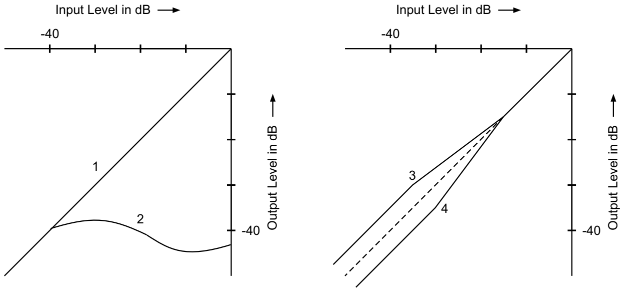

Figure 6: *Compression characteristic of Dolby A. Curve 2: characteristic of the control signal added to the direct path. This results in the compression (3) and expansion (4) characteristics.*

levels (High: -30 dB, Medium: -48 dB and Low: -62 dB). Compression takes place only below these input levels. In the High and Medium setting the frequency band is divided at 800 Hz by a sliding band filter. In the Low setting only frequencies above 800 Hz are affected. The time constants of the analysis modules (Modulation Control Circuits A,B,C) are adapted to the psychoacoustics of hearing. The system is capable to modify the signal spectrum in five bands depending on the frequency content of the input signal.

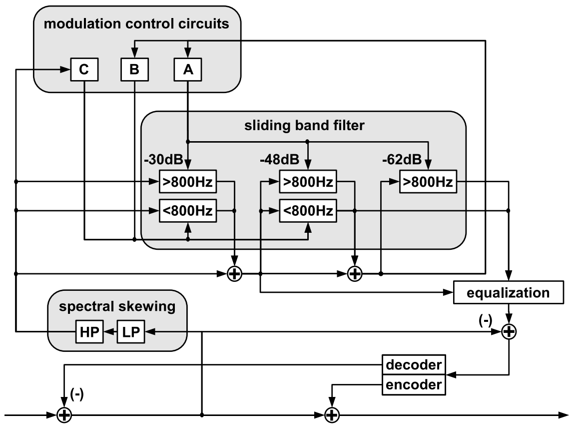

Figure 7: *Block diagram of the Dolby SR compander system [3].*

3.2. TELEFUNKEN Telcom C4

The high end studio compander system from TELEFUNKEN was called Telcom C4. A block diagram is shown in Figure 8. This compander uses a compression ratio of 2/3 working in four different bands separated at 215 Hz, 1450 Hz and 4800 Hz. The compression characteristic is shown in Figure 9.

It shows different compression onsets and thresholds for each frequency band and more compression at higher frequency bands. In the low frequency bands higher levels are expected and thus the compression onset is set to 0 dB whereas at the 10 kHz frequency band a much lower threshold yields a higher gain of up to 25 dB at low levels. The envelope detection method of the Telcom C4 compander uses three time constants: attack, slow release and fast release ($1/10^{th}$ of the slow release time) after a 25 ms duration of

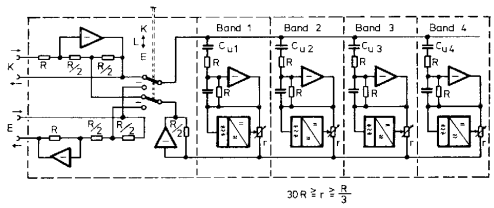

Figure 8: *Block diagram of the TELEFUNKEN Telcom C4 compander system [3].*

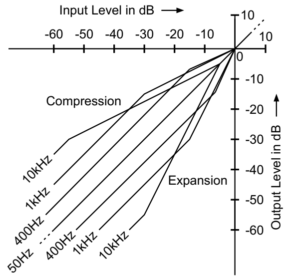

Figure 9: *Compression characteristic of the TELEFUNKEN Telcom C4 compander system [3].*

the slow release time interval. This behavior is illustrated in Figure 4, where the temporal behavior of different envelope estimates according to attack and release time constants is shown.

3.3. Dolby B and C

Dolby B and C (see Figure 10) are the single sliding band consumer derivatives of the professional systems. The B type acts on frequencies only above 500 Hz. Depending on the signal energy this cut-off frequency moves higher. A maximum noise reduction of 20 dB between 2 kHz and 10 kHz can be achieved with this version. Dolby C is a two step version derived from Dolby SR. It also has a constant spectral skewing and improves the high frequency dynamic recording level (reduction of overload inclination). The maximum noise reduction here is approximately 15 dB. The compression characteristics are shown in Figure 11.

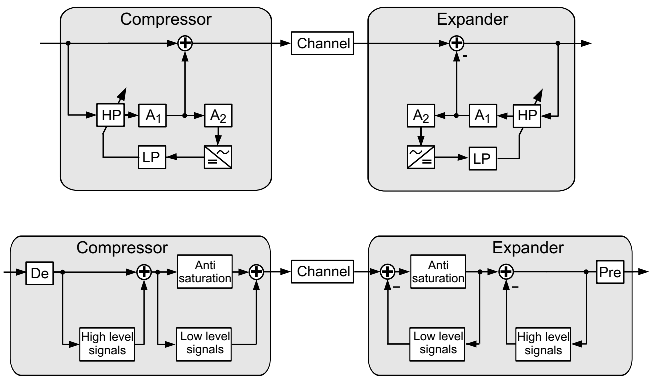

Figure 10: *Block diagram of the Dolby B (top) and Dolby C (bottom) compander systems [3].*

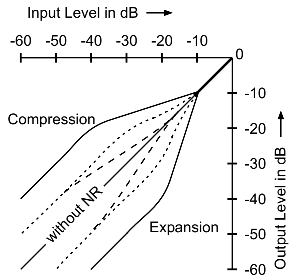

Figure 11: *Compression characteristic of Dolby C [3].*

3.4. TELEFUNKEN HighCOM

Figure 12 shows the block-diagram of the semi-professional consumer version of the Telcom C4 compander. It is a single-band feed-back version with pre- and de-emphasis from 1.2 kHz to 8.6 kHz, a compression factor of 2 and different compression thresholds (see Figure 13).

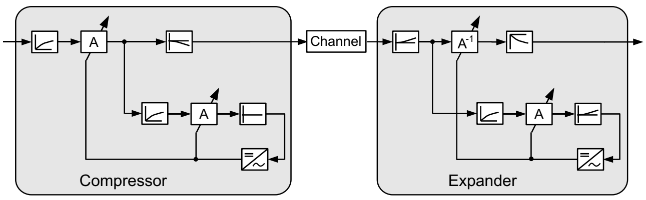

Figure 12: *Block diagram of the TELEFUNKEN HighCOM single-band consumer version of the professional Telcom C4 system [3].*

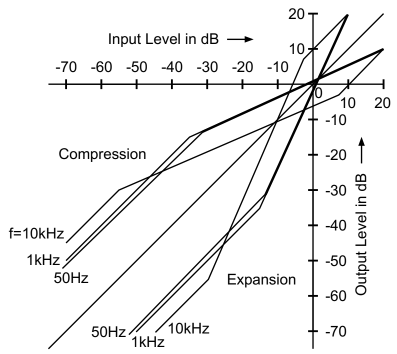

Figure 13: *Compression characteristic of HighCOM [3].*

3.5. dbx and Burwen Noise Eliminator

The dbx compander (Figure 14) was used for professional and consumer tape recording. It is a single band multiplicative feed-back compander with a compression ratio of 2 and an envelope extraction according to the true-RMS principle. It also incorporates pre- and de-emphasis from 370 Hz to 1.59 kHz to avoid high frequency overload, the level detector loop contains a low-pass filter with 10 kHz corner frequency and an additional pre-emphasis from 440 Hz to 4.8 kHz. A similar system is the Burwen Noise Eliminator which uses the same characteristics but with a compressor threshold of -70 dB. Both compression characteristics are plotted in Figure 15. The various types of companders use either multiplicative or additive main signal path modification. The compression ratios vary from 2/3 to 2. Professional systems use multi-band companding to better adapt the compression parameters to the psychoacoustics of hearing. Multi-band companding is realized in fixed band or sliding band technique with overlapping and non overlapping band-pass sections. As envelope (level) estimators peak detector and true RMS detectors are used with different attack and decay concepts which influence the audibility of breathing and pumping.

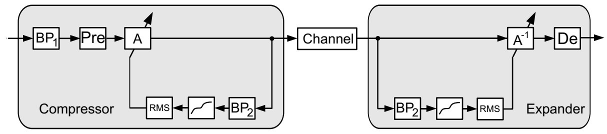

Figure 14: *Block diagram of the dbx compander system [3].*

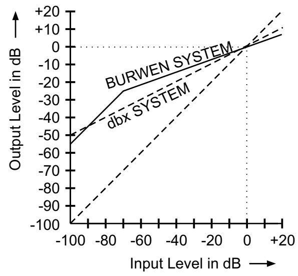

Figure 15: *Compression characteristic of the dbx and the Burwen compander [3].*

4. DIGITAL EMULATION OF ANALOG COMPANDERS

Most of the above mentioned analog systems have been implemented digitally in C-code. This comprises single-band and multi-band structures with peak, modified-peak, RMS and Hilbert level detectors. Adjustable compression factors and adjustable time constants as well as adjustable compression thresholds and compression onsets in each frequency band have been implemented. The block-diagram of the single-band compander emulation is shown in Figure 16.

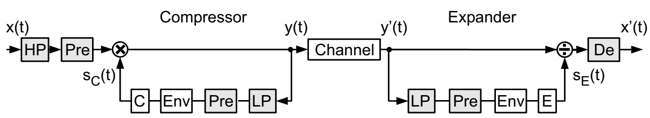

Figure 16: *Block diagram of the digitally implemented feed-back single-band compander emulation. HP/LP: high/low-pass, Pre: pre-emphasis, Env: envelope detection, C/E: compression/expansion factor calculation.*

With this structure several existing types (e.g. Sennheiser HiDynPlus, dbx and the HighCOM) can be re-evaluated. In the single-band version pre- and de-emphasis filters account for the

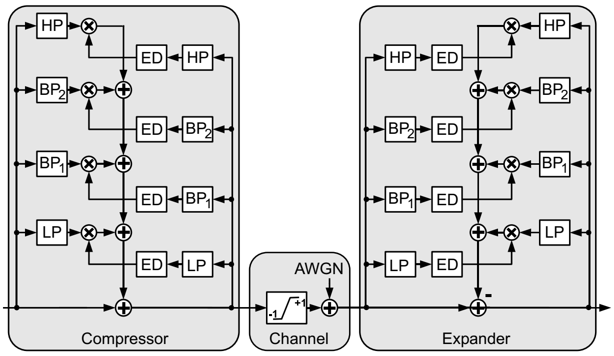

Figure 17: *Block diagram of the digitally implemented feedback multi-band additive compander emulation. HP/LP/BP: high/low/band-pass, ED: envelope detection, AWGN: additive white Gaussian noise.*

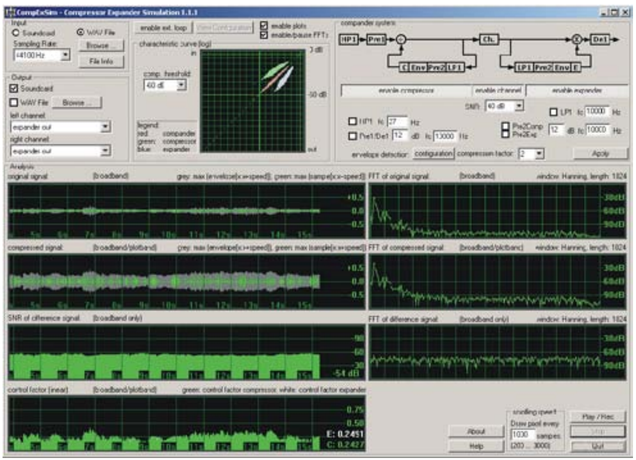

Figure 18: *GUI of the compander emulation program.*

perceptive properties of the ear. All filters have been realized as shelving filters according to [4]. Pre-emphasis in the control path reduces the high frequency dynamic range. To overcome this problem an additional LP filter can be added to the control path. In order to get even better performance for professional usage we also implemented a multi-band system. Dolby A or SR structures have not been implemented because of their well known disadvantages in FM applications.

The multi-band system is shown in Figure 17 and is based on shelving and peak filter designs introduced in [4]. By using this multi-band structure, equation (1) holds and the transfer function of the compressor $H_C(z,t)$ and expander $H_E(z,t)$ can be expressed as

$$H_C(z,t) = 1 + H_{FB}(z,t) \quad (11)$$

$$H_E(z,t) = \frac{1}{1 + H_{FB}(z,t)}. \quad (12)$$

$H_{FB}(z,t)$ denotes the time-varying transfer function of the filter bank. It depends on the signal spectrum of the compressed signal. Compression factors, time constants and envelope estimators define the time dependency of the transfer function. These parameters control the envelope detection and the compression factor calculation in block ED in Figure 17. The control path is separated from the signal path. Both paths work with the same filter bank transfer function. The recursive structure in the expander has delay free loops and is calculated by the technique introduced in [5]. A limiter and a AWGN channel is used between compressor and expander to simulate an analog FM transmission link.

All compander algorithms are written in float or 24 bit integer resolution and are controlled by a GUI offering different IO paths (.wav-file and sound card) and analyzer options. The C++ program can be linked via soundcard to a real-time FM modulator and demodulator. The aim of this program is to have a basis to do high quality audio test sessions to investigate acoustically optimal compander structures and compander settings for wireless FM microphone systems. Figure 18 shows the GUI of the compander evaluation program [6]. It allows to change all compander settings during playback of the test materials. Audio test material can be processed in real-time. Measurements of total harmonic distortion, intermodulation distortion, on/offset behavior and dynamic tracking can be performed for different compander structures and settings. All results are comparable to the analog circuitry. The multi-band compander outperforms the single-band compander. With a high compression ratio in the upper frequency bands and different compression onsets and thresholds a channel noise with a SNR of 40 dB can perceptively be suppressed by the multi-band compander.

5. CONCLUSIONS

The digital implementation of a multi-band compander is suitable to work as a high-end compander system for professional wireless FM microphone systems. The real-time PC implementation of the novel multi-band compander structure allows the possibility to optimize the compander presets by artists and sound engineers. The possibility to integrate an analog FM link into the signal chain offers the evaluation of the compander performance, increases the reliability of the compander settings also under real operating conditions, and also reduces the development time for the audio signal processing significantly.

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