TECHNICAL DOCUMENTATION FOR MUTIL-CHANNEL COMPRESSION HEARING AIDS FOR HEARING AID APPLICATIONS

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INTRODUCTION

The Multi-Channel Audio Compression is designed for multi-channel compression for hearing aid in real time. The document provides detailed description of the implementation of this algorithm. The source codes of this document are all written in C/C++ programming language. Figure 1 shows the pipeline of audio compression algorithm. The input signal is $sig$ and the output/processed signal is $opwav$. All the source codes can be found in our website.

![Fig 1. Pipeline of Multi-Channel Audio Compression](image)

- In Block A, there are several pre-processing stages for input signal including an elliptic high pass filter (remove infrasonic) and a levelling based on rms.
- In Block B, there are 10 gains (Insertion gains) corresponding to 10 frequencies (125Hz to 10000Hz) and based on these the input signal is processed by filter.
- In Block C, N-channels AGC (automatic gain control) filter banks which are designed are used to compress. In our source codes we are using 5 channels.
- In Block D, we can adjust the output gain.
- The parameters including $sig$, $eq\_sig$, $proc\_sig$ and $opwav$ are same as the parameters in the source code.
- In Design a gain function using IG65 block, we design the filter to provide the gain coefficients which is used in block B.
- The Create Multi-Channels filter banks provide the channel parameters and filter bank coefficients which is used in block C.
1. Pre-Processing

1.1 High Pass Filter

**Goal:** The High pass filter is an elliptical filter which is mainly used to remove the infrasonic sounds i.e. Frequencies less than 50Hz.

**How to apply:** For initial processing, `Loadfile` function is called in the main file as shown in the Figure 2. Other Parameters like `framesize` and `Frequency` are also provided along with the input data frame – `sig_frame`.

```c
sig_filt_out = m_loadfile(main_parameters->main_variables, sig_frame, framesize, Fs);
```

Figure 2: LoadFile Function

The Elliptic High Pass Filter is of order 3, the `initial_variables` are used for initialization of the frame processing. The `ellip()` function is used to retrieve the predesigned IIR High Pass Elliptic Filter coefficients as shown in the Figure 3.

```c
m_variables = initial_variables(FrameSize); // FrameSize Used here
ellip(m_variables, 3, 0.1, 0.1, 40 * 1.22 / (Fs / 2), 3);
iiirhp(m_variables->hpf8, m_variables->hpfA, sig, zst, sig_filt, zo, FrameSize);
```

Figure 3: Initialization and IIR High Pass Filtering

1.2 RMS

**Goal:** In this stage, RMS value of the input audio signal is corrected to `ipfiledigrms`, taking into account the gaps between sounds i.e.by ignoring them. This is presumed to be equivalent to a 65 dB input level.

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How to apply: The Input Frame passes through the RMS (Root Mean Square) Stage as shown in the figure 4. The RMS of the Signal is calculated for every RMSFrameLength. Currently this parameter is fixed to evaluate the RMS for every 5 seconds of Data.

```c
//Update RMS for every RMSFrameLength
if (RMSFrameCounter == RMSFrameLength - 1)
{
    for (int k = 0; k < framesize; k++)
        sig_filt_rms[RMSFrameCounter*framesize + k] = sig_frame[k];

    RMSFrameCounter = 0;
    RMSUpdate = 1;
}
else
{
    for (int k = 0; k < framesize; k++)
        sig_filt_rms[RMSFrameCounter*framesize + k] = sig_frame[k];
    RMSUpdate = 0;
    RMSFrameCounter++;
}

if (RMSUpdate)
{
    RMSUpdate = 0;
}

rms = channel_rms(sig_filt_rms, framesize \ RMSFrameLength, Fs, dB_rel_rms);
```

Figure 4: RMS Stage

Channel RMS of the 5 seconds of Input Signal is calculated in 2 stages. Figure 5 shows the 1st stage.

```c
for (int i = 0;i<len_s;i++)
    sum_sig_square += pow(sig[i],2);

first_stage_rms = sqrt(sum_sig_square/len_s);
```

Figure 5: First Stage Channel RMS

To generate 2nd Stage RMS, a parameter every_dB is calculated as shown in Figure 6 which is used to construct histogram.
A histogram is constructed by sorting the frames in descending order with respect to value of the parameter \textit{every\_dB} and only those percentage of frames which have larger \textit{every\_dB} is considered for calculation of the Second Stage RMS. This is shown in Figure 7.

The RMS is used to determine the ratio, which is used to scale the Incoming input signal as shown in Figure 8. All these operations are carried out in block A.
2. Gain Function

2.1 Prescription Design

**Goal:** In *Design a gain function using IG65* block, the insert gains and the frequencies are processed in the Prescription design function to calculate the gain coefficients which are used in Block B.

**How to get the coefficients:**

The Gain coefficients are calculated offline by changing the insertion gains in the frequency band which is part of the Prescription Design function. Figure 9 shows the snapshot of the code where the user can change the insertion gains corresponding to the frequencies to get the 132 order filter coefficients.

```c
//prescription design function
double *ig_eq;
ig_eq = (double *)malloc(134 * sizeof(double));
float freq[10] = { 125, 250, 500, 1000, 2000, 3000, 4000, 6000, 8000, 10000 };
float gain[10] = { 0, 3, 5, 10, 15, 20, 25, 30, 30, 30 };
int span_msec = 6;
ig_eq = prescription_design(freq, fs, gain, span_msec);
```

Figure 9: Prescription Design Function

These gain coefficients are then used in the recalculate stage in the real-time pipeline.

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2.2 Recalculate

**Goal:** Based on the gain coefficients calculated in the prescription design stage, the input frame processed by filter.

**How to apply:** The scaled input from block A is then fed to the block B (recalculate stage), which performs FIR Filtering using \textit{IG_EQ\_Filter} FIR Coefficients which are predetermined in Prescription Design stage as shown in figure 10.

```c
double *m_recalculate(double *sig, int len_s)
{
    double *eq_sig = firFrameFilt(sig, IG_EQ_Filter, RecalFilterBuffer, len_s, RecalFilterSize);

    return eq_sig;
}
```

Figure 10: Recalculate Stage

### 3. Filter Banks

#### 3.1 Multi-channel Filter bank

**Goal:** To design 5 Channel FilterBank and Obtain Channel Specific DualChannel AGC Parameters like \texttt{dig\_chan\_lvl\_0dBgain} and \texttt{dig\_chan\_dBthrs and recomdB}

**How to apply:** Using the coefficients of prescription design and the Band Pass filter coefficients, additional Dual Channel AGC parameters which are specific to each channel are calculated. These parameters are calibrated using the \textit{calibratefilterbankoverlap} function which is shown in Figure 11.
The parameters `chan0dBgn_lvl` and `chan_dBthr` are used as `dig_chan_lvl_0dBgain` and `dig_chan_dBthrs` in the Dual Channel AGC Function and `Calib_recomb_dBpost` is used as `recombdB` in the `NChanFBankAGCAid` function.

### 3.2 Filter banks and Dual Channel AGC

**Goal:** Using preset attack/release times and other specific channel parameters, process the signal in five filter banks.

**How to apply:**

Based on Desired Simulation Level of Hearing Aid, The Input Signal is rescaled with `re_level`. Figure 12 is the snapshot of the above process.

```c
for (int i = 0; i < framesize; i++)
    eg_sig_re[i] = eg_sig[i] * re_level;
```

Figure 12: Rescaling Input Frame with `re_level`

The signal is then processed through a Multi-Channel FilterBanks - `NChanFBankAGCAid` comprised of 5 filters followed by a `DualChannelAGC` in each channel. All the filtered output is time aligned by compensating for the shifts in each frame, this function is shown in figure 13.
for (ix = 1; ix <= nchans; ix++)
{
    bpf_len = (int)bpf[ix - 1][0];
    for (j = 2; j <= bpf_len + 1; j++)
    {
        bpf_chan[j - 2] = bpf[ix - 1][j - 1];
    }
}
#endif BRIANMOORE
int alignFrameLen = (bpf_len_max - bpf_len) / 2;
#endif

double *sigframeout = FrameBasedFilterBank(signal, bpf_chan, FrameSize, bpf_len, ix - 1);
#endif BRIANMOORE
for (int ii = 0; ii < alignFrameLen; ii++)
    alignFrame[ii] = alignBuffer[ix - 1][ii];

for (int ii = 0 ; ii < (FrameSize - alignFrameLen); ii++)
alignFrame[ii + alignFrameLen] = sigframeout[ii];
for (int ii = 0; ii < (alignFrameLen); ii++)
    alignBuffer[ix - 1][ii] = sigframeout[ii + (FrameSize - alignFrameLen)];
#endif

Figure 13: Multi-Channel FilterBanks -NChanFBankAGCAid Stage

Dual Channel AGC system uses two different gain control systems to protect the user from uncomfortable loudness produced by brief intense sounds. Based on the ANSI standards, the attack and the release time and the channel parameters, and the limiter parameters, the envelope is weighted by the gain.

for (i = 0; i < framesize; i++)
{
    gain_env[i] = pow(env[i], exponent) * g0dB; // Multiplying with Normalized gain to get 0dB gain from compressor for channel
    // printf("%.2f ", gain_env[i]);
}

int *limit = (int*)calloc((FrameSize + 1), (sizeof(int)));
for (i = 0; i < framesize; i++)
{
    if (envlim[i] > env[i])
    { limit[i] = i;
    // printf("%.16f ",limit[j]);
    j++;
}

Figure 14: Dual Channel AGC Compressor Envelope

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Here is the relevant part of the envelope which is multiplied by the normalized gain. This is shown in figure 14:

```c
for (i = 0; i < envlim_size; i++)
{
    gain_envp[(int)(limit[i])] = gain_envp[(int)(limit[i])] * (gain_lim[i]); //Limiter Gain Applied
    //printf("%f ", gain_envp[1000]);
}

//spc_act = round(10000 * length(limit) / length(gain_envlp)) / 100; % activity measure, to 0.01% NOT ENCODED
int nshift;
for (i = 0; i < framesize; i++)
{
    proc[i] = (double*)malloc((framesize + 1), (sizeof(double)));
    //printf("%d ", proclen);
    double *ip = (double*)malloc((framesize + 1), (sizeof(double)));

    for (i = 0; i < FrameSize; i++)
    {
        proc[i] += wx1[i] * pow(10, 0.05 * recombdB[ix - 1]);
        wx1[i] = 0;
    }
}
```

Figure 15: Dual Channel AGC Limiter Envelope

The channel limiter gain is applied. This operation is done on all 5 channels for an input frame by appropriately applying specific channel parameters. This is shown in in figure 15.

The Output from `DualChannelAGC` is weighted with the parameter `recombdB`, which is based on the channel number and added with rest of the channel outputs. This is shown in figure 16.

```c
final_gain = pow(10, 0.05*(opfiledigrms - ipfiledigrms));
for (int i = 0; i < framesize; i++)
    opwav_frame[i] = proc[i] * final_gain;
```

Figure 17: Final Gain Scaling for the Signal Frame

4. **Gain Limiter**

**Goal:**
Change the volume of output signal to the desired volume based the SPL (sound pressure level).

**How to apply:**
The output frame is finally multiplied with final gain. Figure 17 shows the gain scaling.