DREAM 1.2.4

Auto Frequency Acquisition:

Clicking on the input spectrum plot changes the mixing frequency for demodulation. If the Auto Frequency Acquisition is enabled, the largest peak near the curser is selected.

PLL:

The Phase-Lock-Loop (PLL) tracks the carrier of the modulated received signal. The resulting phase offset between the reference oscillator and the received carrier is displayed in a dial control. If the pointer is almost steady, the PLL is locked. If the pointer of the dial control turns quickly, the PLL is out of lock. To get the PLL locked, the frequency offset to the true carrier frequency must not exceed a few Hz.

Carrier Frequency:

The (estimated) carrier frequency of the analog signal is shown. (The estimation of this parameter can be done by the Autom Frequency Acquisition which uses the estimated PSD of the input signal and applies a maximum search.)

Save Audio as WAV:

Save the audio signal as stereo, 16-bit, 48 kHz sample rate PCM wave file. Checking this box will let the user choose a file name for the recording.

Mute Audio:

The audio can be muted by checking this box. The reaction of checking or unchecking this box is delayed by approx. 1 second due to the audio buffers.

Demodulation Type:

The following analog demodulation types are available:
AM:

This analog demodulation type is used in most of the hardware radios. The envelope of the complex base-band signal is used followed by a high-pass filter to remove the DC offset. Additionally, a low pass filter with the same bandwidth as the pass-band filter is applied to reduce the noise caused by non-linear distortions.

LSB / USB:

These are single-side-band (SSB) demodulation types. Only one side of the spectrum is evaluated, the upper side band is used in USB and the lower side band with LSB. It is important for SSB demodulation that the DC frequency of the analog signal is known to get satisfactory results. The DC frequency is automatically estimated by starting a new acquisition or by clicking on the plot.

CW:

This demodulation type can be used to receive CW signals. Only a narrow frequency band in a fixed distance to the mixing frequency is used. By clicking on the spectrum plot, the center position of the band pass filter can be set.

FM:

This is a narrow band frequency demodulation.

Filter Bandwidth:

A band-pass filter is applied before the actual demodulation process. With this filter, adjacent signals are attenuated. The bandwidth of this filter can be chosen in steps of 1 Hz by using the slider bar. Clicking on the right or left side of the slider leveler will increase/decrease the bandwidth by 1 kHz.
The current filter bandwidth is indicated in the spectrum plot by a selection bar.

AGC (Automatic Gain Control):

Input signals can have a large variation in power due to channel impairments. To compensate for that, an automatic gain control can be applied. The AGC has four settings: Off, Slow, Medium and Fast.

Noise Reduction:

The noise suppression is a frequency domain optimal filter design based algorithm. The noise PSD is estimated utilizing a minimum statistic. A problem of this type of algorithm is that it produces the so called "musical tones". The noise becomes colored and sounds a bit strange. At the same time, the useful signal (which might be speech or music) is also distorted by the algorithm. By selecting the level of noise reduction, a compromise between distortion of the useful signal and actual noise reduction can be made.

Under the GNU General Public License (GPL)

 Open-Source Software
Implementation of a DRM-Receiver1.2.4cvs<center>Dream, Version
</p>
FFTW <i>http://www.fftw.org</i>
FAAD2
<i>AAC/HE-AAC/HE-AACv2/DRM decoder (c) Ahead Software, www.nero.com
(http://faac.sf.net)</i>
FAAC
<i>http://faac.sourceforge.net</i>
QWT <i>Dream is based in part
on the work of the Qwt project (http://qwt.sf.net).</i>
Hamlib
<i>http://hamlib.sourceforge.net</i>
FhG IIS Journaline Decoder
<i>Features NewsService Journaline(R) decoder technology by Fraunhofer IIS,
Erlangen, Germany. For more information visit
http://www.iis.fhg.de/dab</i>
FreeImage <i>This software uses the
FreeImage open source image library. See http://freeimage.sourceforge.net for
details. FreeImage is used under the GNU
GPL.</i>

<hr/>

<center>CREDITS</center>
We
want to thank all the contributors to the Dream software (in alphabetical
order):

Developers<center><p>Bakker,
Menno</p><p>Cesco</p><p>Fillod, Stephane</p><p>Fine, Mark J.</p><p>Manninen,
Tomi</p><p>Pascutto, Gian C.</p><p>Richard, Doyle</p><p>Russo,
Andrea</p></center>
Parts of Dream are based on code
by<center><p>Karn, Phil (www.ka9q.net)</p><p>Ptolemy Project
(http://ptolemy.eecs.berkeley.edu)</p><p>Tavernini, Lucio
(http://tavernini.com/home.html)</p><p>The Math Forum
(http://mathforum.org)</p><p>The Synthesis ToolKit in C++ (STK)
(http://ccrma.stanford.edu/software/stk)</p></center>
Supporters<cente
r><p>Jose de Amorim, Roberto</p><p>Kainka, Burkhard</p><p>Keil,
Jens</p><p>Kilian, Gerd</p><p>Knuetter, Carsten</p><p>Ramisch,
Roland</p><p>Schall, Norbert</p><p>Schill, Dietmar</p><p>Schneider,
Klaus</p><p>Stöppler, Simone</p><p>Varlamov, Oleg</p><p>Wade,
Graham</p></center>
</p>
<p></big></p>
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You should have received a copy of the GNU General Public License along with his program; if not, write to the Free Software Foundation, Inc., 59 Temple Place, Suite 330, Boston, MA 02111-1307 USA</p>
<p><font color="#ff0000" face="courier"><p><big>Dream is a software implementation of a Digital Radio Mondiale (DRM) receiver. All what is needed to receive DRM transmissions is a PC with a sound card and a modified analog short-wave (MW, LW) receiver.Although this software is going to be distributed as free software under the terms of the GPL this does not mean that its use is free of rights of others. The use may infringe third party IP and thus may not be legal in some countries.This compilation of Dream uses the following libraries:1OnHelpAbout()&About...1OnHelpWhatsThis()What's &ThisSound

Waterfall Display of Input Spectrum:

The input spectrum is displayed as a waterfall type. The different colors represent different levels.<b>

SNR History:

The history of the values for the SNR and correctly decoded audio blocks is shown. The maximum achievable number of correctly decoded audio blocks per DRM frame is 10 or 5 depending on the audio sample rate (24 kHz or 12 kHz AAC core bandwidth).

Doppler / Delay History:

The history of the values for the Doppler and Impulse response length is shown. Large Doppler values might be responsable for audio drop-outs.

Frequency Offset / Sample Rate Offset History:

The history of the values for frequency offset and sample rate offset estimation is shown. If the frequency offset drift is very small, this is an indication that the analog front end is of high quality.

Audio Spectrum:

This plot shows the averaged audio spectrum of the currently played audio. With this plot the actual audio bandwidth can easily determined. Since a linear scale is used for the frequency axis, most of the energy of the signal is usually concentrated on the far left side of the spectrum.

Input PSD:

This plot shows the estimated power spectral density (PSD) of the input signal. The PSD is estimated by averaging some Hamming Window weighted Fourier transformed blocks of the input signal samples. The dashed vertical line shows the estimated DC frequency.

Input Spectrum:

This plot shows the Fast Fourier Transformation (FFT) of the input signal. This plot is active in both modes, analog and digital. There is no averaging applied. The screen shot of the Evaluation Dialog shows the significant shape of a DRM signal (almost rectangular). The dashed vertical line shows the estimated DC frequency. This line is very important for the analog AM demodulation. Each time a new carrier frequency is acquired, the red line shows the selected AM spectrum. If more than one AM spectrums are within the sound card frequency range, the strongest signal is chosen.

SNR Spectrum (Weighted MER on MSC Cells):

This plot shows the Weighted MER on MSC cells for each carrier separately.

Shifted PSD:
This plot shows the estimated Power Spectrum Density (PSD) of the input signal. The DC frequency (red dashed vertical line) is fixed at 6 kHz. If the frequency offset acquisition was successful, the rectangular DRM spectrum should show up with a center frequency of 6 kHz. This plot represents the frequency synchronized OFDM spectrum. If the frequency synchronization was successful, the useful signal really shows up only inside the actual DRM bandwidth since the side loops have in this case only energy between the samples in the frequency domain. On the sample positions outside the actual DRM spectrum, the DRM signal has zero crossings because of the orthogonality. Therefore this spectrum represents NOT the actual spectrum but the "idealized" OFDM spectrum.

FAC, SDC, MSC:

The plots show the constellations of the FAC, SDC and MSC logical channel of the DRM stream. Depending on the current transmitter settings, the SDC and MSC can have 4-QAM, 16-QAM or 64-QAM modulation.<b>

Transfer Function / Group Delay:

This plot shows the squared magnitude and the group delay of the estimated channel at each sub-carrier.

Impulse Response:

This plot shows the estimated Impulse Response (IR) of the channel based on the channel estimation. It is the averaged, Hamming Window weighted Fourier back transformation of the transfer function. The length of PDS estimation and time synchronization tracking is based on this function. The two red dashed vertical lines show the beginning and the end of the guard-interval. The two black dashed vertical lines show the estimated beginning and end of the PDS of the channel (derived from the averaged impulse response estimation). If the "First Peak" timing tracking method is chosen, a bound for peak estimation (horizontal dashed red line) is shown. Only peaks above this bound are used for timing estimation.

This is the official logo of the Dream software.

Service Selectors:

In a DRM stream up to four services can be carried. The service type can either be audio, data or audio and data. If a data service is selected, the Multimedia Dialog will automatically show up. On the right of each service selection button a short description of the service is shown. If a service is an audio and data service, a "+ MM" is added to this text. If a service is an audio and data service and this service is selected, by opening the Multimedia Dialog, the data can be viewed while the audio is still playing.

Station Label and Info Display:

In the big label with the black background the station label and some other information about the current selected service is displayed. The red text on the top shows the audio compression format (e.g. AAC), the sample rate of the core coder without SBR (e.g. 24 kHz), if SBR is used and what audio mode is used (mono, stereo, P-stereo -> low-complexity or parametric stereo). In case SBR is used, the actual sample rate is twice the sample rate of the core AAC decoder.

The next two types of information are the language and the program type of the service (e.g. German / News).

The big turquoise text in the middle is the station label. This label may appear later than the red text since this information is transmitted in a different logical channel of a DRM stream.

The turquoise text on the bottom shows the gross bit-rate in kbits per second of the current selected service. The abbreviations EEP and UEP stand for Equal Error Protection and Unequal Error Protection. UEP is a feature of DRM for a graceful degradation of the decoded audio signal in case of a bad reception situation. UEP means that some parts of the audio is higher protected and some parts are lower protected (the ratio of higher protected part length to total length is shown in the brackets). On the right, the ID number connected with this service is shown.

Status LEDs:

The three status LEDs show the current CRC status of the three logical channels of a DRM stream. These LEDs are the same as the top LEDs on the Evaluation Dialog.

Input Level:

The input level meter shows the relative input signal peak level in dB. If the level is too high, the meter turns from green to red. The red region should be avoided since overload causes distortions which degrade the reception performance. Too low levels should be avoided too, since in this case the Signal-to-Noise Ratio (SNR) degrades.<b>

Text Message:

On the top right the text message label is shown. This label only appears when an actual text message is transmitted. If the current service does not transmit a text message, the label will be invisible.

Show &all stations1OnShowStationsMenu(int)Show &only active stationsC:\cvs\drm\common\GUI-QT\StationsDlg.cppLanguageSiteCountryPower [kW]TargetTime [UTC]Station Name%02d:%02d UTCftp://216.92.35.131/DRMSchedule.iniDream Schedule UpdateDream tries to download the newest DRM schedulefrom www.drm-dx.de (powered by Klaus Schneider).Your computer must be connected to the internet.The current file DRMSchedule.ini will be overwritten.Do you want to continue?Update successful.Due to network problems with the Windows version of QT, the Dream software must be restarted after a DRMSchedule update.Please exit Dream now.OkUpdate failed. The following things may caused the failure:- the internet connection was not set up properly- the server www.drm-dx.de is currently not available- the file 'DRMSchedule.ini' could not be writtenThe file AMSchedule.ini could not be found or contains no data.No stations can be displayed.The file DRMSchedule.ini could not be found or contains no data.No stations can be displayed.Try to download this file by using the 'Update' menu.%04d-%04d?=Rig unknown setting: Rig set parm failed: Rig set func failed: Rig set level failed: Rig set mode failed: Rig open failed.Rig set conf failed.Malformatted config string.Initialization of hamlib failed.No rig model ID selected.

Signal-Meter:

Shows the signal strength level in dB relative to S9.
Note that not all front-ends controlled by hamlib support this feature. If the s-meter is not available, the controls are disabled.

UTC Time:

Shows the current Coordinated Universal Time (UTC) which is also known as Greenwich Mean Time (GMT).

Frequency Counter:

The current frequency value can be changed by using this counter. The tuning steps are 100 kHz for the buttons with three arrows, 10 kHz for the buttons with two arrows and 1 kHz for the buttons having only one arrow. By keeping the button pressed, the values are increased / decreased automatically.

Stations List:

In the stations list view all DRM stations which are stored in the DRMSchedule.ini file are shown. It is possible to show only active stations by changing a setting in the 'view' menu. The color of the cube on the left of a menu item shows the current status of the DRM transmission. A green box shows that the transmission takes place right now, a yellow cube shows that this is a test transmission and with a red cube it is shown that the transmission is offline.
If the stations preview is active an orange box shows the stations that will be active.
The list can be sorted by clicking on the headline of the column.
By clicking on a menu item, a remote front-end can be automatically switched to the current frequency and the Dream software is reset to a new acquisition (to speed up the synchronization process). Also, the log-file frequency edit is automatically updated.

Interferer Rejection:

There are two algorithms available to reject interferers:

Bandpass Filter (BP-Filter):

The bandpass filter is designed to have the same bandwidth as the DRM signal. If, e.g., a strong signal is close to the border of the actual DRM signal, under some conditions this signal will produce interference in the useful bandwidth of the DRM signal although it is not on the same frequency as the DRM signal. The reason for that behaviour lies in the way the OFDM demodulation is done. Since OFDM demodulation is a block-wise operation, a windowing has to be applied (which is rectangular in case of OFDM). As a result, the spectrum of a signal is convoluted with a Sinc function in the frequency domain. If a sinusoidal signal close to the border of the DRM signal is considered, its spectrum will not be a distinct peak but a shifted Sinc function. So its spectrum is broadened caused by the windowing. Thus, it will spread in the DRM spectrum and act as an in-band interferer.
There is a special case if the sinusoidal signal is in a distance of a multiple of the carrier spacing of the DRM signal. Since the Sinc function has zeros at certain positions it happens that in this case the zeros are exactly at the sub-carrier frequencies of the DRM signal. In this case, no interference takes place. If the sinusoidal signal is in a distance of a multiple of the carrier spacing plus half of the carrier spacing away from the DRM signal, the interference reaches its maximum.
As a result, if only one DRM signal is present in the 20 kHz bandwidth, bandpass filtering has no effect. Also, if the interferer is far away from the DRM signal, filtering will not give much improvement since the squared magnitude of the spectrum of the Sinc function is approx -15 dB down at 1 1/2 carrier spacing (approx 70 Hz with DRM mode B) and goes down to approx -30 dB at 10 times the carrier spacing plus 1 / 2 of the carrier spacing (approx 525 Hz with DRM mode

B). The bandpass filter must have very sharp edges otherwise the gain in performance will be very small.

Modified Metrics:

Based on the information from the SNR versus sub-carrier estimation, the metrics for the Viterbi decoder can be modified so that sub-carriers with high noise are attenuated and do not contribute too much to the decoding result. That can improve reception under bad conditions but may worsen the reception in situations where a lot of fading happens and no interferer are present since the SNR estimation may be not correct.

Chart Selector:

With the chart selector different types of graphical display of parameters and receiver states can be chosen. The different plot types are sorted in different groups. To open a group just double-click on the group or click on the plus left of the group name. After clicking on an item it is possible to choose other items by using the up / down arrow keys. With these keys it is also possible to open and close the groups by using the left / right arrow keys.
A separate chart window for a selected item can be opened by right click on the item and click on the context menu item.

Received time - date:

This label shows the received time and date in UTC. This information is carried in the SDC channel.

Number of Services:

This shows the number of audio and data services transmitted in the DRM stream. The maximum number of streams is four.

Prot. Level (B/A):

The error protection level of the channel coder. For 64-QAM, there are four protection levels defined in the DRM standard. Protection level 0 has the highest protection whereas level 3 has the lowest protection. The letters A and B are the names of the higher and lower protected parts of a DRM block when Unequal Error Protection (UEP) is used. If Equal Error Protection (EEP) is used, only the protection level of part B is valid.

SDC / MSC Mode:

Shows the modulation type of the SDC and MSC channel. For the MSC channel, some hierarchical modes are defined which can provide a very strong protected service channel.

Interleaver Depth:

The symbol interleaver depth can be either short (approx. 400 ms) or long (approx. 2 s). The longer the interleaver the better the channel decoder can correct errors from slow fading signals. But the longer the interleaver length the longer the delay until (after a re-synchronization) audio can be heard.

DRM Mode / Bandwidth:

In a DRM system, four possible robustness modes are defined to adapt the system to different channel conditions. According to the DRM standard:
<i>

Mode A:

Gaussian channels, with minor fading

Mode B:

Time and frequency selective channels, with longer delay spread

Mode C:

As robustness mode B, but with higher Doppler spread

Mode D:

As robustness mode B, but with severe delay and Doppler spread
The bandwith is the gross bandwidth of the current DRM signal

SNR:

Signal to Noise Ratio (SNR) estimation based on FAC cells. Since the FAC cells are only located approximately in the region 0-5 kHz relative to the DRM DC frequency, it may happen that the SNR value is very high although the DRM spectrum on the left side of the DRM DC frequency is heavily distorted or disturbed by an interferer so that the true overall SNR is lower as indicated by the SNR value. Similarly, the SNR value might show a very low value but audio can still be decoded if only the right side of the DRM spectrum is degraded by an interferer.

First Peak:

This algorithms searches for the first peak in the estimated impulse response and moves this peak to the beginning of the guard-interval (timing tracking algorithm).

Guard Energy:

Time synchronization tracking algorithm utilizes the estimation of the impulse response. This method tries to maximize the energy in the guard-interval to set the correct timing.

Channel Estimation Settings: With these settings, the channel estimation method in time and frequency direction can be selected. The default values use the most powerful algorithms. For more detailed information about the estimation algorithms there are a lot of papers and books available.

DFT Zero Pad: Channel estimation method for the frequency direction using Discrete Fourier Transformation (DFT) to transform the channel estimation at the pilot positions to the time domain. There, a zero padding is applied to get a higher resolution in the frequency domain -> estimates at the data cells. This algorithm is very speed efficient but has problems at the edges of the OFDM spectrum due to the leakage effect.

Channel Estimation Settings: With these settings, the channel estimation method in time and frequency direction can be selected. The default values use the most powerful algorithms. For more detailed information about the estimation algorithms there are a lot of papers and books available.

Linear: Simple linear interpolation method to get the channel estimate. The real and imaginary parts of the estimated channel at the pilot positions are linearly interpolated. This algorithm causes the lowest CPU load but performs much worse than the Wiener interpolation at low SNRs.

Channel Estimation Settings: With these settings, the channel estimation method in time and frequency direction can be selected. The default values use the most powerful algorithms. For more detailed information about the estimation algorithms there are a lot of papers and books available.

Wiener: Wiener interpolation method uses estimation of the statistics of the channel to design an optimal filter for noise reduction.
Freq: In this edit control, the current selected frequency on the front-end can be specified. This frequency will be written into the log file.

Log File: Checking this box brings the Dream software to write a log file about the current reception. Every minute the average SNR, number of correct decoded FAC and number of correct decoded MSC blocks are logged including some additional information, e.g. the station label and bit-rate. The log mechanism works only for audio services using AAC source coding. During the logging no Dream windows should be moved or re-sized. This can lead to incorrect log files (problem with QT timer implementation under Windows). This problem does not exist in the Linux version of Dream.
The log file will be written in the directory were the Dream application was started and the name of this file is always DreamLog.txt

Reverberation Effect: If this check box is checked, a reverberation effect is applied each time an audio drop-out occurs. With this effect it is possible to mask short drop-outs.

Flip Input Spectrum: Checking this box will flip or invert the input spectrum. This is necessary if the mixer in the front-end uses the lower side band.

MLC, Number of Iterations: In DRM, a multilevel channel coder is used. With this code it is possible to iterate the decoding process in the decoder to improve the decoding result. The more iterations are used the better the result will be. But switching to more iterations will increase the CPU load. Simulations showed that the first iteration (number of iterations = 1) gives the most improvement (approx. 1.5 dB at a BER of 10-4 on a Gaussian channel, Mode A, 10 kHz bandwidth). The improvement of the second iteration will be as small as 0.3 dB.
The recommended number of iterations given in the DRM standard is one iteration (number of iterations = 1).

MSC CRC LED: This LED shows the status of the Main Service Channel (MSC). This channel contains the actual audio and data bits. The LED shows the CRC check of the AAC core decoder. The SBR has a separate CRC, but this status is not shown with this LED. If SBR CRC is wrong but the AAC CRC is ok one can still hear something (of course, the high frequencies are not there in this case). If this LED turns red, interruptions of the audio are heard. The yellow light shows that only one 40 ms audio frame CRC was wrong. This causes usually no hearable artifacts.

SDC CRC LED: This LED shows the CRC check result of the Service Description Channel (SDC) which is one logical channel of the DRM stream. This data is transmitted in approx. 1 second intervals and contains information about station label, audio and data format, etc. The error protection is normally lower than the protection of the FAC. Therefore this LED will turn to red earlier than the FAC LED in general.
If the CRC check is ok but errors in the content were detected, the LED turns yellow.

FAC CRC LED: This LED shows the Cyclic Redundancy Check (CRC) of the Fast Access Channel (FAC) of DRM. FAC is one of the three logical channels and is always modulated with a 4-QAM. If the FAC CRC check was successful, the receiver changes to tracking mode. The FAC LED is the indication whether the receiver is synchronized to a DRM transmission or not.

The bandwidth of the DRM signal, the constellation scheme of MSC and SDC channels and the interleaver depth are some of the parameters which are provided by the FAC.

Frame Sync LED: The DRM frame synchronization status is shown with this LED. This LED is also only active during acquisition state of the Dream receiver. In tracking mode, this LED is always green.Time Sync Acq LED: This LED shows the state of the timing acquisition (search for the beginning of an OFDM symbol). If the acquisition is done, this LED will stay green.

I / O Interface LED: This LED shows the current status of the sound card interface. The yellow light shows that the audio output was corrected. Since the sample rate of the transmitter and local computer are different, from time to time the audio buffers will overflow or under run and a correction is necessary. When a correction occurs, a "click" sound can be heard. The red light shows that a buffer was lost in the sound card input stream. This can happen if a thread with a higher priority is at 100% and the Dream software cannot read the provided blocks fast enough. In this case, the Dream software will instantly loose the synchronization and has to re-synchronize. Another reason for red light is that the processor is too slow for running the Dream software.

Doppler / Delay: The Doppler frequency of the channel is estimated for the Wiener filter design of channel estimation in time direction. If linear interpolation is set for channel estimation in time direction, this estimation is not updated. The Doppler frequency is an indication of how fast the channel varies with time. The higher the frequency, the faster the channel changes are.
The total delay of the Power Delay Spectrum (PDS) is estimated from the impulse response estimation derived from the channel estimation. This delay corresponds to the range between the two vertical dashed black lines in the Impulse Response (IR) plot.

Sample Frequency Offset: This is the estimation of the sample rate offset between the sound card sample rate of the local computer and the sample rate of the D / A (digital to analog) converter in the transmitter. Usually the sample rate offset is very constant for a given sound card. Therefore it is useful to inform the Dream software about this value at application startup to increase the acquisition speed and reliability.

DC Frequency Offset: This is the estimation of the DC frequency offset. This offset corresponds to the resulting sound card intermedia frequency of the front-end. This frequency is not restricted to a certain value. The only restriction is that the DRM spectrum must be completely inside the bandwidth of the sound card.

1OnButtonStartStop()C:\cvs\drm\common\GUI-QT\TransmDlg.cppDream DRM TransmitterThis is a test transmissionFull PathSize [KB]File Name16-QAM4-QAMnew&StopImage Files (*.png *.jpg *.jpeg *.jfif)MSC interleaver mode: The symbol interleaver depth can be either short (approx. 400 ms) or long (approx. 2 s). The longer the interleaver the better the channel decoder can correct errors from slow fading signals. But the longer the interleaver length the longer the delay until (after a re-synchronization) audio can be heard.

Output format:

Since the sound-card outputs signals in stereo format, it is possible to output the DRM signal in three formats:
real valued output on both, left and right, sound-card channels
I / Q output which is the in-phase and quadrature component of the complex base-band signal at the desired IF. In-phase is output on the left channel and quadrature on the right channel.

E / P output which is the envelope and phase on separate channels. This output type cannot be used if the Dream transmitter is regularly compiled with a sound-card sample rate of 48 kHz since the spectrum of these components exceed the bandwidth of 20 kHz.
The envelope signal is output on the left channel and the phase is output on the right channel.
Output intermediate frequency of DRM signal: Set the output intermediate frequency (IF) of generated DRM signal in the 'sound-card pass-band'. In some DRM modes, the IF is located at the edge of the DRM signal, in other modes it is centered. The IF should be chosen that the DRM signal lies entirely inside the sound-card bandwidth.DRM Bandwidth: The bandwith is the gross bandwidth of the generated DRM signal. Not all DRM robustness mode / bandwidth constellations are possible, e.g., DRM robustness mode D and C are only defined for the bandwidths 10 kHz and 20 kHz.

DRM Robustness Mode: In a DRM system, four possible robustness modes are defined to adapt the system to different channel conditions. According to the DRM standard:
<i>
Mode A:</i> Gaussian channels, with minor fading
<i>
Mode B:</i> Time and frequency selective channels, with longer delay spread
<i>
Mode C:</i> As robustness mode B, but with higher Doppler spread
<i>
Mode D:</i> As robustness mode B, but with severe delay and Doppler spread

Input Level: The input level meter shows the relative input signal peak level in dB. If the level is too high, the meter turns from green to red

```
 [option] [argument]

Recognized options:
-t, --transmitter           DRM transmitter mode
-p, --flipspectrum          flip input spectrum
-i <n>, -
-mlciter <n>       number of MLC iterations                              allowed
range: 0...4                               default: 1
-s <r>, --sampleoff <r>     sample rate offset initial value [Hz]
allowed range: -200.0...200.0
-m, --muteaudio             mute audio output
-f <s>, --fileio <s>        disable sound card,                              use
file <s> instead
-w <s>, --writewav <s>      write output to wave file
-S <r>, --fracwinsize <r>   freq. acqu. search window size [Hz]
-E <r>, --fracwincent <r>   freq. acqu. search window center [Hz]
-F, --filter                apply bandpass filter
-D, --modmetric             enable modified metrics
-c <n>, --inchansel <n>     input channel selection
0: left channel
1: right channel
2: mix both channels (default)
3: I / Q input positive
4: I / Q input negative
5: I / Q input positive (0 Hz IF)
6: I / Q input negative (0 Hz IF)

-u <n>, --outchansel <n>    output channel selection
0: L -> L, R -> R (default)
1: L -> L, R muted
2: L muted, R -> R
3: mix -> L, R muted
4: L muted, mix -> R
```

```
-r <n>, --frequency <n>     set frequency [kHz] for log file
-a <s>, --latitude <s>      set latitude string for log file
-o <s>, --longitude <s>     set longitude string for log file
-l <n>, --startlog <n>      start log file (delayed by
<n> seconds)                              allowed range: 1...3600
-y <n>, --

colorscheme <n>   set color scheme for main plot

0: blue-white (default)

1: green-black

2: black-grey  --mdioutadr <s>

MDI out network address                               format [IP#]:[port]  --
mdiinport <n>             set MDI in port number
-I <n>, --snddevin <n>      set sound in device
-O <n>, --snddevout <n>     set sound out device
-M <n>, --hamlib-model <n>  set Hamlib radio model ID
-C <s>, --hamlib-config <s> set Hamlib config parameter
-T, --ensmeter              enable S-Meter
-h, -?, --help             this help textExample: Usage: '' needs a string
argument' needs a numeric argument between  and ]IRQ
```