Audio processing and ALC in the FT-897D

I recently bought an FT-897D, and after a period of operation noticed problems with what I perceived to be a low average level of output power and reports of muffled audio. This is the story of my investigation to improve the audio quality and increase the average level of transmitted power.

From the start I assumed that the main problem of muffled speech was going to be a mismatch between my voice, which seems to have a 800Hz peak in the spectrum, and the Yaesu MH-31 microphone. Selecting position 1 and 2 (low cut) on the rear of the microphone made some difference to the overall quality. The position 2 improved intelligibility but seemed to reduce the average transmitted power still further.

Experiments followed with Electret microphone capsules (which improved the quality) and Heil HC-4 & HC5 microphone inserts coupled with the OBP speech processor module. All of these were fitted inside the original MH-31 microphone case, but I fairly soon came to the conclusion that the fundamental problem of ‘muffled’ audio was due to the case design, especially the slotted mouthpiece vents which seem to create ‘notches’ in the audio response when the microphone is held at certain positions relative to the mouth.

The next stage was to try a cheap electret boom microphone and headset. This made a big difference to the audio quality, and immediately solved most of the problems I had with the reports of ‘muffled’ speech.

However the average transmitted audio level was still low. I constructed a compressor circuit, which made a huge difference to the average audio level when tested using a PC soundcard and ‘Goldwave’ audio editing software. However when I connected it to the radio it made practically no difference to the transmitted level. I then added a further clipper stage to the output of the compressor. This combination resulted in only 1dB variation of audio level when speaking as all the transient peaks were clipped off. Once again I connected it to the radio, but the average power remained about the same. So what was going on ?

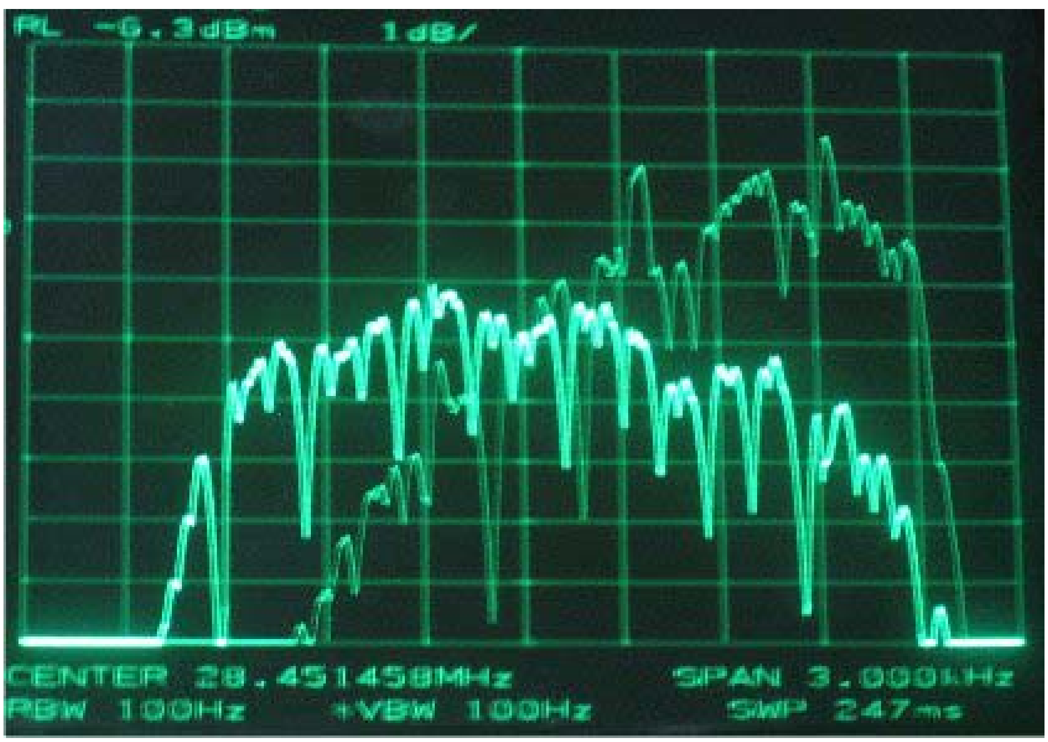
The main problem with compression and clipping, is that when the processed audio is passed through further stages of amplification and filtering, the resultant phase changes can cause the peaks of the audio to be regenerated. Without redesigning the radio circuits to provide a more linear phase response this problem would always remain. I also suspected that the ALC circuit was causing additional difficulties. As I increased the average output power by compressing the audio, the ALC simply decreased the transmitter gain in order to maintain the same average output level.

Up to this point I had been concerned about not overdriving the ALC, and had tried to follow the user manual recommendation that the ALC meter should only indicate a ‘few’ bars on occasional voice peaks.

If you set the mic gain with no processing so that the ALC meter only kicks up to say 4 bars on voice peaks, the ALC will hardly be operating. If you now switch on the processor with a medium setting of say 30 the ALC will kick up quite a lot higher. So should the processor be backed off to maintain the original peaks or perhaps the microphone gain should be reduced ? I decided to try and find the optimum mic gain and processor settings.

The first stage was to measure the variation in gain produced by altering the microphone and processor level controls. I used 'Spectrum Laboratory' software to generate some audio tones and sweeps, and a separate receiver and spectrum analyser to monitor the demodulated audio and RF spectrum. This revealed some surprises. The first was the two tone output spectrum of the radio. Which although within the published specification, looked worse that I was expecting. Varying output power control and varying the input level from no ALC to full ALC didn't make a huge difference to the spectrum within +/- 20KHz of the centre frequency. This is to be expected with static tones, as the ALC simply reduces the overall gain in order to maintain the correct output level.

At this point everything started to become clear. The mic gain had a range of about 25dB when varied from 5 to 100, and perhaps more interestingly, the processor gain control also had a similar range, the only real difference being a slight amount of additional filtering which reduced the bass content and gave a 2-3dB boost to frequencies around the 2 to 2.5KHz region.



The above picture shows the effect of switching the processor on. The radio has been fed with a slow audio sweep, and the Spectrum analyser scale has been arranged to show 3KHz bandwidth, with the centre corresponding to 1.5KHz. The processing gain has been set to 15 which give approximately the same audio gain at 1.5KHz. The audio level has been set so that it is well below the point of ALC operation. The bright trace is with the processor switched off, the other trace with it switched on. No DSP filtering has been selected. Ignore the lumps and bumps in the response and just concentrate on the overall trend.

This frequency response is very similar to that produced by the Heil HC-4 microphone which is renowned for its ‘punch’. So the processing function seems to work by simply boosting the level of audio to bring on a greater degree of ALC action, resulting in compression on speech peaks.

I decided to measure the relative gain of the mic and processor level controls. In order to obtain these figures had to fix a reference point, so I set each control in turn to maximum and reduced the audio level to give a 10w output (in order to keep well under the point of ALC operation) with a 1KHz sinusoidal tone. I then decreased the control setting and measured the increase in audio level required to produce the original level of output power. This produced reliable and repeatable results (within a few dB margin of measurement error). This permits calculation of the relative gain associated with each control setting, as per the following results.

SETTING	Mic Gain WRT position 5	Processor Gain WRT position 5	Processor Gain WRT 0dB gain
100	+25dB	+25dB	+17dB
90	+24dB	+24dB	+16dB
80	+23dB	+23dB	+15dB
70	+22dB	+21dB	+13dB
60	+21dB	+20dB	+12dB
50	+20dB	+19dB	+11dB
40	+18dB	+17dB	+9dB
30	+16dB	+14dB	+6dB
25	+14dB	+13dB	+5dB
20	+12dB	+11dB	+3dB
15	+10dB	+8dB	0dB
10	+6dB	+5dB	-3dB
5	0dB	0dB	-8dB
0	N/A	N/A	N/A

Note that setting the processor gain to 15 results in approximately the same audio level as with the processor switched off. The fourth column shows the processor gain relative to this setting.

Following these measurements I tested the range of ALC operation. The ALC meter indicates the approximate levels of gain reduction as shown below:-

ALC Meter reading (just peaking to	Calculated gain reduction

bar)	
9+	Greater than 14dB
9	14dB
8	9dB
7	4dB
6	2dB
5	1dB
4	0.5dB
3	0dB
2	0dB
1	0dB

If you set the ssb mic gain to indicate just a 1 or 2 bars on speech peaks, this point is just before the onset of ALC action.

The next test was to measure the output power level on speech using a spectrum analyser with an averaging function. This produced the following results:-

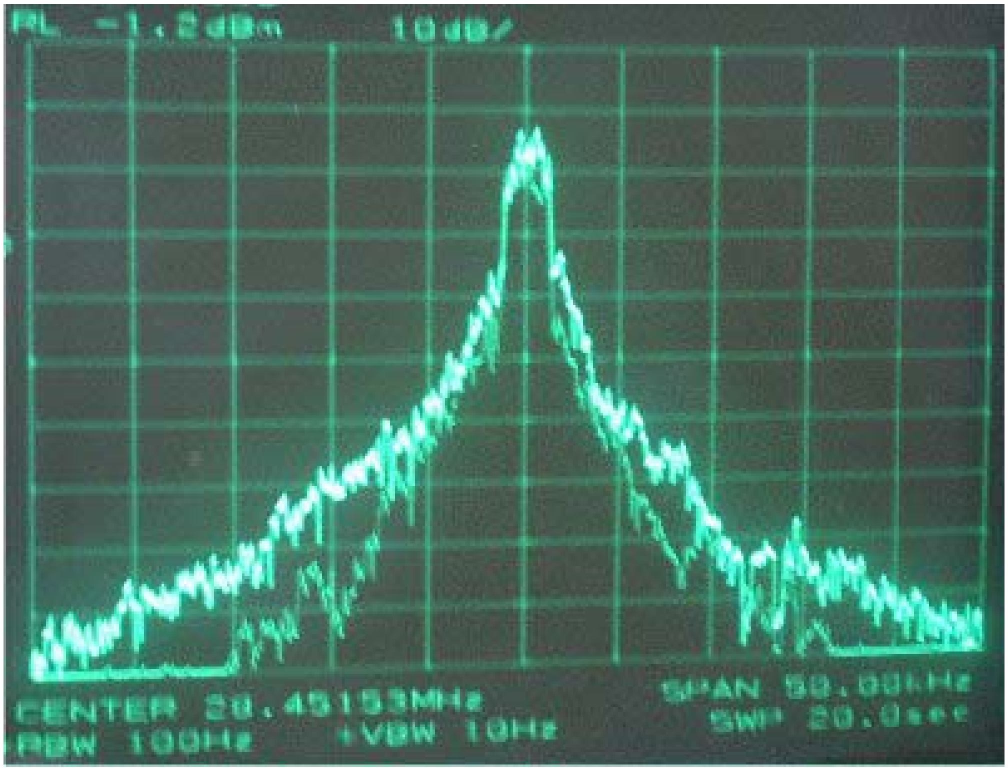
ALC meter reading on speech (just peaking to bar)	Approximate average power WRT single tone at full output
9+	-4dB typical (up to -3dB)
9	-6dB
8	-8dB
7	-9dB
Less than 7	-10dB or less

Note that these results are only guidance figures, and depend upon to the combination of voice and microphone being used. The levels are only very approximate and it is not possible to achieve full output power unless a single sinusoidal tone is used to modulate the transmitter. However it can be seen that even with modest ALC action the average output power can be quite low. By increasing the mic gain or processing level so that the ALC peaks to 9 rather than 3 or 4 bars, an increase in average output power of greater than 3dB is possible.

However the problem with too much ALC is that the bandwidth of the transmitted signal can increase, due to generation of additional intermodulation products outside the wanted signal. This can result in ‘splatter’ or interference to other operators on adjacent frequencies. The ALC action ‘modulates’ the ssb signal resulting in the generation of additional sidebands around the wanted signal. This effect is not evident in the normal two tone lab tests used for intermodulation products, because it does not dynamically vary the level of ALC. Varying the output power control does not have much effect either, as the ALC circuit still remains in operation. A very good description of ALC operation and the problems it can cause has been written by SM 5 BSZ. Titled - The abominable ALC it can be found at:-

http://www.nitehawk.com/sm5bsz/dynrange/alc.htm

The following picture shows the effect of modulating the FT-897D with white noise (to simulate speech) and capturing the resulting signal with a spectrum analyser using the max hold function.



The bright trace shows the radio with maximum mic gain and processing and full ALC operation (worst case). The dim trace shows low mic gain with no processing, just before the onset of ALC. The top graticule line represents 100W CW power, the span is 50KHz so each division represents 5KHz. Note that the difference in peak levels of the wanted signal are only a few dB's but the unwanted skirts are quite a bit higher.

Thrashing the ALC circuit results in a 20dB higher level of 'noise' at 10KHz away from the centre frequency, and 10dB higher level of 'noise' at 15KHz offset.

In order to get this is perspective, you have to consider that this is likely to be the worst case, and that as the sidebands are already approx 60 – 70dB down on carrier, so an extra few dB increase in adjacent channel noise due to ALC operation is unlikely to cause many complaints. The radio has been built to a price and for casual use works very well. Unless you are using a very good antenna, external linear amplifier, or the band is very quiet and you have other stations operating in close proximity, it is likely to go unnoticed. I am not suggesting that this is good practise, but when compared to the output spectrum with no ALC operation, it's not too much worse.

Although all these notes refer to my experiences with the FT-897D, most manufacturers now seem to rely upon using the ALC circuit to provide some form of speech processing. Some models are work better than others, and the performance cannot simply be determined by using static two tone intermodulation tests. The best method of maximising the output power would be to provide better control of audio levels prior to the modulation process, with the ALC circuit being used to prevent occasional overdriving the PA on speech peaks.

As it is not easy to disable the ALC and use an external speech processor, what is the best compromise in terms of control settings on the FT-897D ?

I suggest the following:-

If you are using the standard Yaesu MH-31 microphone, switch the mic to position 2, which removes a low frequency peak and flattens the overall frequency response. Changing the transmit and receive carrier offsets can also make a big difference to speech clarity, you can also experiment with the DSP settings, although I found that this didn't make much difference to my speech.

Set the mic gain so that the ALC meter is just kicking up to 3 or four bars on speech peaks. Do not change the mic gain once you have found this level. This setting is ideal for local contacts or rag chewing on 80m.

If you wish to sound a bit louder, switch the processor on and set it to 15, which shouldn't increase the overall audio level, but will give the 2 to 2.5KHz speech frequencies a slight boost, improving intelligibility and making it sound as though you are using a Heil HC-4 microphone capsule at a fraction of the price.

If the going gets tough and you wish to have a bit more 'punch' turn the processor up to about 30 or 40. This will give about 6-9dB gain compression which will produce about 2dB increase in average power with causing too many problems on adjacent channels. Turning the processor up to 80 will produce a further 1 dB increase in average power, but at the expense of much more 'splatter' onto adjacent frequencies. I would not suggest increasing the processor control any more than this as distortion increases dramatically and speech intelligibility actually worsens.

If you wish to improve the clarity of speech the cheapest method seems to be to swap the microphone insert for an electret capsule. This made a huge difference to my speech quality and only cost £1.50.

Although these notes specifically refer to the FT-897D, they are equally applicable to most of the current generation of amateur radio transceivers. I hope you find them useful. Please feel free to copy the content providing you acknowledge the author.

Martin Ehrenfried, G8JNJ, 09 Sept 2007 V1.0