IRCA Technical Column

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A home-built double-superhet LW/MW receiver with sync AM-detection

by Ad Dieleman

1 Introduction

When I read about synchronous AM-detection for the first time in the 1970's, I immmediately realized that this was the way to go to properly receive amplitude-modulated signals and in 1983 I started to build a general-coverage receiver for 150 kHz-30 MHz around a sync-detector. However, I never got around to it; I built it up to the 2nd mixer stage (see the block diagram in Figure 1) and then I built a MW-only front-end, because MW has always been my primary interest in radio. I put up a front-end with a local oscillator and two ganged RF tuned circuits and RF amplifier, a conventional approach for a MW-receiver. And it worked reasonably well, I receive some Americans at my place in Dordrecht, Holland every now and then. But after a while I got fed up with its varying sensitivity and the fact that I never got the ganging between LO and RF circuits right.

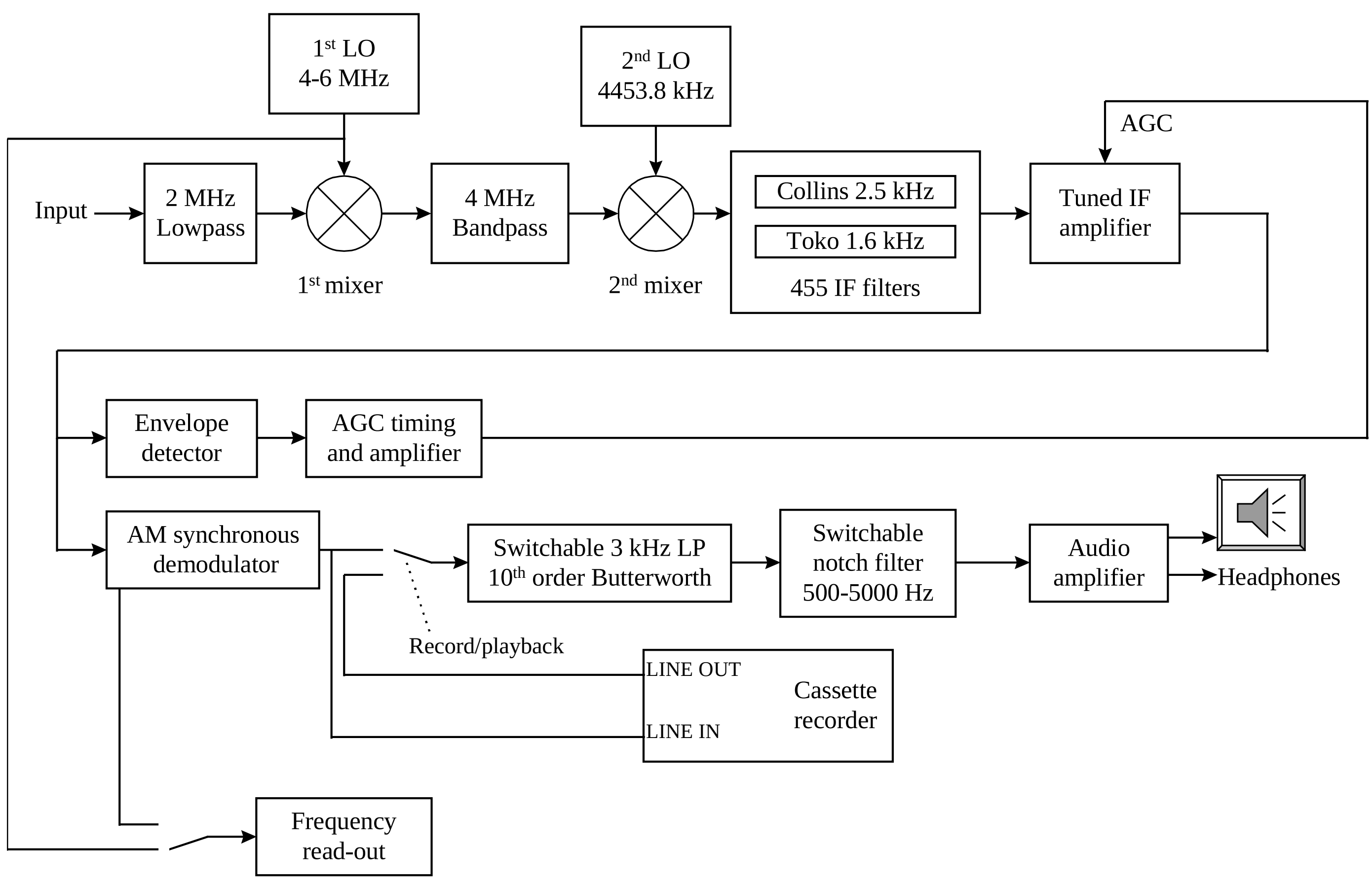

Figure 1 Receiver Block Diagram

I decided to follow a different approach: "Why not make a MW-receiver with a high 1st IF as is usual in 0.1-30 MHz HF-receivers?" In such an HF-receiver the first IF is placed above the highest frequency to be received, resulting in a first IF of 45 up to 75 Mhz in various designs. Making the first local oscillator for such an IF is a major undertaking: obtaining adequate stability and freedom of sideband noise in a frequency range well above 50 MHz is not easy at all. Only with elaborate synthesizer circuitry and/or direct digital synthesis is this possible and still only the best receivers reportedly offer good performance in this respect.

However, for a MW-only receiver the first IF theoretically can be placed just above 1.7 MHz, being the highest MW frequency presently in use. 2 MHz seems a practical lower limit with good filtering before the first mixer. I chose 4 MHz because filtering requirements are more relaxed, the relative variation of the first LO frequency is less and, most important of all, my frequency read-out IC counts from 0-3999.9 and then reverts to 0000.0 for 4 MHz. Directly feeding the LO-signal of 4-6 MHz results thus in the received-signal frequency read-out of 0-2 MHz. Sometimes you need this kind of luck… Building a good local oscillator with a range of 4-6 MHz can be done with an LC-oscillator, which is what I did. Stability proves just adequate and sideband noise is not the limiting factor for the dynamic range of my receiver.

In the following description I shall only discuss the principles laid down in the block diagram. Circuit diagrams are not presented, partly because I don't want to fill a whole year's editions of DX Monitor, but also because I want to avoid the suggestion that you can build it by simply following the circuit diagrams.

2 From the input filter to the first IF

Returning to the block diagram you will see that the input signal is filtered by a lowpass-filter with a cut-off at 2 MHz. It is a nine-pole (4 L's, 5 C's) Chebishev-filter with a 1 dB ripple in the passband and an impedance of 500 ohms to match the input impedance of the mixer stage. The 1st mixer is built up around a Plessey SL6440. In earlier days I used an LM1496 at its highest allowable current and with a feedback resistor, this worked almost just as well!

The local oscillator is an LC-oscillator; it uses a large-size inductor and a Jackson 100 pF variable capacitor and its tuning range is 4-6 MHz. The reduction drive unit was taken from an R107 army radio. The receiver therefore has a tuning range of 0-2 MHz. In practice you can tune down to 3 kHz or so and I receive a lot of signals from 10 kHz upwards. To mimimize drift during warm-up I used a BF240 bipolar transistor as the active element; it only needs 1 mA to perform well. It is followed by a buffer of a JFET/bipolar cascode to drive the mixer and another emitter-follower for the counter output. It drifts about 500 Hz in the first few hours after switch-on and then it drifts a few 100 Hz between day and night in accordance with the room temperature. This is good enough for listening, but not for unattended recording. If there is one thing to improve in my radio, it's long-term oscillator stability.

The resulting signal is filtered at 4 MHz. This "roofing filter" consists of two LC-bandfilters, separated by a low-gain amplifier. The input signal to the second mixer is about 2x that of the first mixer. With a conservative estimate of 50 for the Q of the bandfilters this gives a minimum image rejection for the second IF of 105 dB, which is amply sufficient. Its bandwidth is of the order of 50 kHz, so this means that the second mixer still gets a fairly large amount of signals to process.

3 The second IF circuitry

The 2nd mixer is again a Plessey SL6440. The 2nd local oscillator is a JFET-based crystal-oscillator. The frequency looks a bit odd; In the old days of 1988 I chose 453.8 kHz as IF because this was the centre frequency of the Toko IF-filter. Again a happy coincidence here: 453.8 kHz is near perfect as a carrier frequency to receive an AM-signal with its carrier as a USB-signal through the 2.5 kHz Collins-filter. I had the crystal made by KVG on custom order.

Then the IF-filters. There is a choice of two bandwidths "wide" and "narrow", switched with 2 relays, one at the input of the filters and one at the output. I didn't dare to use one relay because I was afraid of input-to-output cross-talk. The relays ground both input and output of the filter that is not in use. For the narrow bandwidth, intended for tough DX in a crowded band, I took a Toko mechanical filter of 1.6 kHz wide. Sadly enough, it disappointed me then and it still does. For SSB AM-listening with sync detection you need to include the carrier in the passband. The carrier must not be attenuated more than about 15 dB, otherwise my sync circuit begins to have difficulties acquiring lock. With the Toko's bandwidth of 1.6 kHz, the resulting audio sounds too dull to be well-readable. Even in true SSB-fashion I'm not comfortable with the audio bandwidth.

For the "wide" position I opted for an LFD-4 ceramic filter, being 4 kHz wide (manufacturer? nothing on the filter!). Its shape factor is very good, less than 2, but it still proved too wide for DX-work. Tuning the carrier on the edge provides up to 4 kHz wide USB or LSB reception, which sounds very clear but also sharp. For program listening I usually tuned the carrier halfway from the centre to the edge, resulting in an audio bandwidth of over 3 kHz. This provided a nice, clear and very well-readable sound. Over this holiday period I replaced the LFD-4 "wide" filter with a Collins 2.5 kHz filter, the one that is sold as an add-on for the AR7030. Yes, it was expensive, it cost me 100 Euros! And yes, it makes a big difference. Again I am startled by the great influence of the IF filter selectivity on the receiver's character as a whole. I readily believe that DSP-filtering with its unheard-of shape factors makes for a new experience in DX-listening. Sometime I'll try to have a JRC NRD-545 next to my receiver to really hear the differences.

With the Collins 2.5 kHz filter I can really listen to AM in USB or LSB mode with the carrier locked, with a very well-readable signal and very good selectivity. I'm anxiously awaiting the first real opening to America, at the moment that I'll be awake, that is… I'm not really a die-hard DX-er!

After the filter comes the IF amplifier. It consists of three tuned stages. The first one is a gain-controlled JFET cascode, the other two contain BF981's as the active element. The second stage is also gain-controlled, the third one has a fixed gain. The IF gain is so high, that the AGC just doesn't turn down the gain on the receiver's own noise. I like this behaviour very much: it provides for constant output, no matter how weak the signal is. It is also beneficial for the following sync detector: it gets a constant input signal so that you can optimize lock behaviour for strong and weak signals alike. The tuned stages aid in providing high stopband attenuation; relying on the IF filter alone is a bit risky and not necessary.

4 AGC

There is an envelope detector, but it's only used for AGC! I wasn't ever tempted to use it for signal recovery. The AGC circuit contains an op-amp to provide high loop gain, so that there is very little output variation with signal strength. I never really measured it, but it's well below 3 dB for signals above a few microvolts. A fast-attack, slow-decay circuit is incorporated. For band-scanning there is a reasonably fast mode, there's also a slow mode with a decay time of over 1 second for listening to a fixed station. This is especially helpful when listening to two or more stations on one channel. In that case the carriers interfere and cause the signal strength to vary, while the modulation doesn't in SSB-mode. The slow AGC ignores the strength variations due to the interfering carriers and provides a clean mix of the audio signals. Quite regularly I identify the weaker of two or three stations on one channel this way.

5 Demodulation

The amplified IF-signal is then fed into the synchronous AM-demodulator. I consider this a vital part of my receiver and therefore I'll go somewhat more into detail. In its block diagram a standard PLL is seen with the voltage-controlled oscillator (VCO), phase detector and loop filter/amplifier. The VCO is an LC-oscillator with a varactor providing the voltage control, nothing special here. The linearizing network is incorporated to ensure that the VCO's "gain", i.e. frequency change per input voltage change is about the same for the total frequency range. Linearizing is done with a diode-resistor network around an op-amp.
The loop-filter determines the capture-range and the loop behaviour when locked. It is switchable: in the fast mode the loop bandwidth is about 30 Hz, so that it easily captures while tuning. In the slow mode it's only 1 Hz, providing minimal audio distortion and optimum lock performance. It stays locked for more than 5 seconds in absence of a carrier. The "slow" loop filter is an integrating filter that is trimmed to have no drift in the absence of input signal. With this circuit I'm able to lock effortlessly to carriers of signals of which I can't even hear if they have audio content or not. It happens quite often that I lock to carriers of supposedly transatlantic origin, only to find out that I can't hear any audio but only noise!

The VCO frequency is preset by operating a potentiometer, allowing to position the carrier to lock onto anywhere in the passband of the filter. This effectively creates passband tuning, the frequency read-out however is affected according to the position of the VCO frequency. I quickly got used to this. The frequency read-out can be switched to show the VCO frequency, which is very helpful to pre-set the VCO to USB or LSB reception. For the Collins filter: USB = 453.8 kHz and LSB = 456.2 kHz.

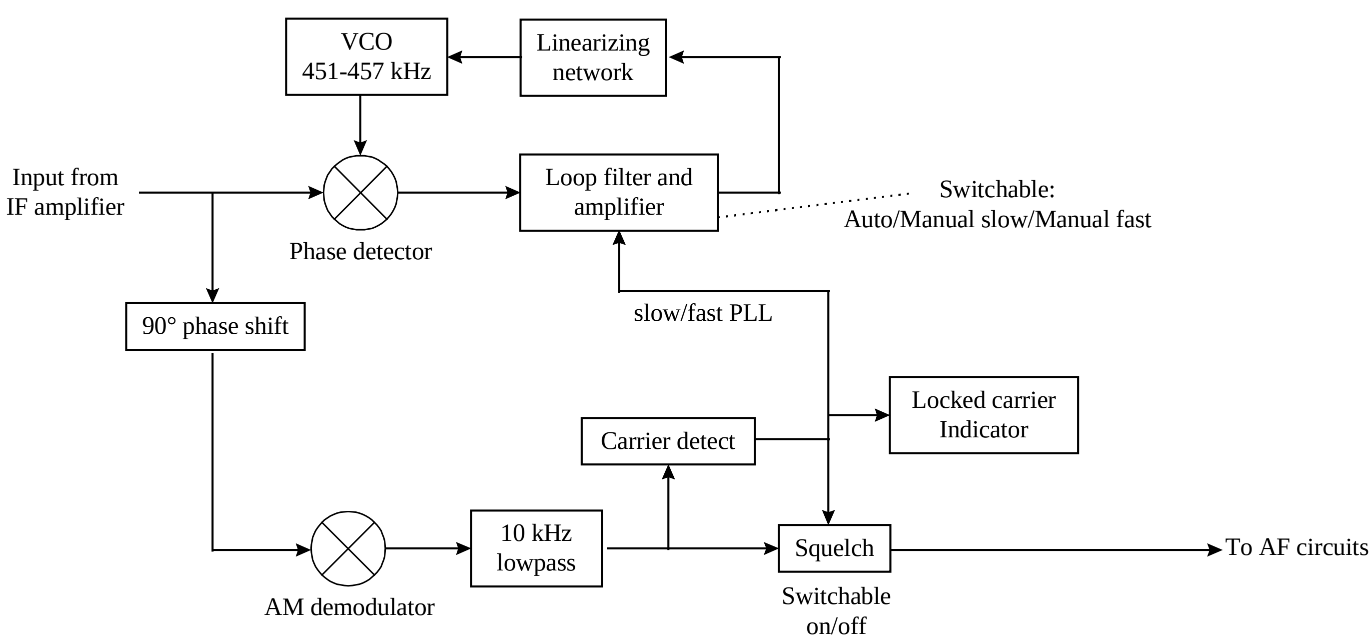

Figure 2 Synchronous AM Demodulator Block Diagram

As you can see I have built in automation in the loop-filter: as soon as lock is detected from the resulting DC-output voltage of the AM-demodulator, the filter switches from fast to slow. In practice I never use it; I just switch manually when I want to. In the fast mode the audio quality is more than good enough to listen to the signal and to decide if you're interested to stay on that frequency. Only then I switch the loop and the AGC to their slow modes and I turn off the squelch to keep hearing the signal during deep fades of the carrier portion of the signal. This is a common occurrence, causing a lot of distortion in envelope detectors!

Both the phase detector and the AM-demodulator are built up with LM1496 double balanced modulators. They always operate in their linear regions. More convenient solutions for PLL-based AM-demodulation exist in single IC's, but this arrangement offers more flexibility in circuit design, allowing all kinds of experiments.

The squelch is open (no muting) when there is sufficient detection voltage, indicating a carrier that the PLL has locked to. If not, the squelch circuit doesn't completely mute, but attenuates the audio signal about 5x (17 dB). This is done to attenuate the otherwise offendingly loud beatnote when approaching a AM station's signal: the carrier is by far the strongest signal in a station's spectrum. The squelch's switch is a CD4053B CMOS analogue switch.

SSB and CW reception is simply done by uncoupling the VCO from the loop and tune it via its potentiometer on the preferred spot of the passband. SSB reception is fine, for CW the filters are of course much too wide.

So far I've found one station that doesn't benefit from synchronous AM reception. It is Europe 1 at 183 kHz. It sounds distorted with sync AM, in both the fast and slow PLL modes as well as on my Sony ICF-SW77 in sync mode. When I switch the Sony to AM envelope detection it sounds normal. It seems like the carrier is phase-modulated with a data signal, but I haven't checked this so far.

6 The audio circuits

First we encounter a filter that can be inserted into the signal path. It is a sharp audio-filter and provides a fixed –3 dB roll-off at 3 kHz and an attenuation of 60 dB at 6 kHz. This represents a shape factor of 2, aiding the radio's selectivity because the sync-detector puts adjacent-channel interference higher up in the spectrum. With the sharp IF-filters used now this audio-filter is hardly ever useful, but sometimes it helps to remove the last bit of interference. The signal's readability suffers from switching in this audio-filter, although the signal's bandwidth is not limited. Remember, the Collins filter is 2.5 kHz wide, which is narrower than the 3 kHz cut-off of the audio filter. My guess is that the non-linear group delay in the audio filter disturbs the signal's coherence.

A audio notch filter is also included. It is based upon a gyrator circuit, this is a circuit that produces inductance behaviour from an electronic circuit and a capacitor. This "inductance" is then combined with a real-world capacitor to make a series LC-circuit filtering out the offending frequency. The notch is about 40 dB deep, alas it is rather wide above 2 kHz, also removing a considerable portion of the wanted signal. Well, it's better than nothing and more than once it helped me out.

The audio-amplifier is constructed from discrete components and is not of the highest possible calibre. I have vague plans to replace it by an IC someday in the coming decade.

7 Using this radio

So what does all this mean when it comes to listening? Of course you have to start by connecting an antenna. I use a home-built tuned loop-antenna with 6 frequency ranges, providing continous coverage from 10 kHz to 4 MHz. After having recovered from writing this article I might write something more about it. This antenna provides a good deal of selectivity and this certainly aids in preventing spurs and birdies in the receiver. I don't have the possibility to set up a beverage aerial, so I can't test it. If someone in Holland or Belgium has one, or an NRD-545 for that matter, I'll happily drive a few 100 km to meet, chat, compare and fiddle around.

Effectively this radio receives broadcast stations in SSB-mode with the aided comfort of perfect and stable zero-beat, allowing one to listen to music without discomfort. Especially the Collins filter provides very good performance this way, although I am a bit irritated by the fact that an adjacent channel sometimes causes interference with a resonance-like sound effect at about 3 kHz. I don't yet know where this comes from, the received signal itself sounds OK and the measured filter characteristic in the radio looks good enough. I know that the filter is quite unforgiving towards improper termination impedances, so the input is terminated with an LC-circuit of 2.2 kOhm resonant impedance and a Q of about 50 and the output is terminated with a 2.2 kOhm resistor.

The receiver handles signals up to about 50-100 mV without protest; above that, extra noise and whistles emerge due to overloading of the SL6440 mixer stages.

Given that I use my loop aerial, I feel that this radio is not the limiting factor in DX-ing. Splatter from strong stations frequently spoil reception of weak stations. I performed a number of tests to see if intermodulation was a source of these disturbances and I had to conclude that this was not the case. The fact that equally strong signals yield very different disturbing effects, bears this out. I can't measure the dynamic range, but I'm guessing that it's somewhere between 80 and 90 dB. So far I found that splatter from nearby strong signals dominates over dynamic range problems in my situation. If you run into dynamic range problems with a dynamic range of 80 dB the transmitter has to have an attenuation of more than 80 dB of its outband signals, otherwise you would hear splatter before the receiver generates intermodulation problems. This can be checked quickly by inserting attenuation: if it's splatter, the interfering noises do not change in strength and character. Therefore a close-in dynamic range of 80-90 dB may well be sufficient; further away, at e.g. 50 kHz distance, the dynamic range has to be larger. In my case the loop antenna's selectivity takes care of that. I speculate that this phenomenon is also the cause of the conclusions in a number of reviews of the JRC NRD-545. The reviewers are in general not enthusiastic about its dynamic range figures when interfering signals are a few kHz away ,and at the same time most reviewers rave about the set's ability to recover weak signals in a crowded band compared to other receivers they have evaluated. Selectivity measurements indicate that the selectivity of the receiver is determined by its filters. The shape factor with the Collins is about 1.9 and stopband attenuation quickly goes up to > 90 dB.

8 The lessons I learned

I think it was a good idea to

- invest in an expensive IF filter. The radio's performance got a giant boost from it.
- opt for the double-superhet concept. It's simpler to build in spite of its more complex block diagram, it offers constant sensitivity and a continous tuning range from VLF to MW.
- choose 455 kHz as a second IF. Of course one could consider building a single-super MW-radio with a high IF. I wouldn't go this route. There is not much to be had in terms of IF-filters above a few MHz; I remember vaguely that Kenwood and/or ICOM used 8-9 MHz crystal IF-filters. The choice in IF-filters is by far the largest at 455 kHz and because the IF filter characteristic is so vital, it's nice to have as many options as possible. It took me three filters to achieve a receiver performance that I feel is right; of course you can go right to a bandwidth of 2.5 kHz with a quality filter and avoid my earlier mistakes.
- put so much effort in a well-working AM sync detector. Don't consider a radio without a well-working synchronous AM-detection if you are serious about broadcast AM listening! Once you've experienced its advantages you won't want to live without it. Having said that, there are a couple of radios around with a badly working sync circuit. Incidentally, I've read that the NRD-545 denies its AGC flexibility to the AM and ECSS listening modes; this is a serious omission in my view.
  If you are planning to build your own sync detector I highly recommend studying Floyd M. Gardner's "Phaselock Techniques" (John Wiley & Sons). It is a thorough and very well-written book on the subject. If you have a good receiver without sync detection, but with an IF output, it surely is an option to buy or build an external AM sync detector unit.
- not put too much effort in attaining a very high dynamic range. So far I haven't felt the necessity to upgrade. I acknowledge that a better antenna or just a different location might necessitate a higher dynamic range, I can't really judge that.
- add a recording input and output. I'm able now to replay that difficult to understand ID or to capture the ID at the whole hour, while I favour doing something else instead of waiting in front of my radio for half an hour only to miss the moment suprème. While I'm at it, even a very good cassette recorder like my Nakamichi CR-4E proved hardly good enough to reproduce the difficult signals without undue tape modulation noise or Dolby pumping effects. I'm seriously considering a mini-disc walkman for this purpose. "Davez" (URL: http://www.ticon.net/~joelt/index.html) is very satisfied with this solution.
- choose 20 V as the main supply voltage. This is high enough to have ample headroom in the various mixing and amplifier stages and low enough to enable usage of almost any active element. Looking back, a voltage of 15 V would still be sufficient, but I wouldn't like to go much lower; 12 V is the absolute lower limit in my view. The consequence of this is that you need two 12 V lead-acid batteries to power the radio, which I did. Forget about up-converting from a single 12 V battery, you'll receive nothing but harmonics of the switching frequency. Believe me, I tried…

It wasn't such a good idea to

- not build a strictly modular receiver. When I would start all over again, I would build each module with appropriate connectors in a sort of rack-mounting set-up.
- build a high-order sharp cut-off audio low-pass filter. It doesn't do much good, a milder filter will undoubtedly do better.
- opt for a two-position only IF filter block. I seriously consider rebuilding it with at least three positions, preferably five or six.

- use multiplexed LED's in the frequency read-out unit. Avoid this like the plague! The kit I bought had this arrangement and it took me a several months of experimenting only to find out that I couldn't eliminate the harmonics of the multiplexing signals out of my radio, let alone out of the nearby loop aerial. I ended up rebuilding the kit by replacing the LEDs by an LCD with the appropriate driver IC's.

Any questions or reactions? You can contact me directly by e-mail: ad.dieleman@planet.nl. I'd love to hear from you!

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