As keith reported in OsmoDevCall on June 11, when an external SIP entity sends a SIP INVITE with SDP for AMR using a dynamic PT, osmo-sip-connector still sends AMR audio using a different (actually unassigned for this call/session) PT.

I guess this relates to the fact that 3GPP uses "pseudo static" PTs for the different codecs, i.e. it has defined that certain PT numbers from the dynamic PT range actually statically represent 3GPP codec types. Within Osmocom, we are sticking to the latter and may not have considered "Real" SIP/SDP compatibility on the external interface.

It's not sipcon sending the payload type number in the AMR audio, it's the osmo-mgw. What is necessary is actually connecting in the first place the codec negotiation in SIP with the one in osmo-msc; hence configuring osmo-mgw correctly.

We have never had proper translation between the codec known in osmo-msc and the codec negotiated by SIP.

As per #1683, the osmo-sip-connector should be transparent about osmo-msc and the codec negotiated by SIP.

I have a set of patches almost ready that rewire MSC's codec figuring, adds SDP via MNCC and makes sipcon transparently forward SDP between external SIP entity and osmo-msc. They are the ominous "codec patches" lying around for years.

I would actually very much like to finish up and merge these patches, because they solve a huge amount of codec issues and were already tested at congress, but so far it's not happening working time wise.