Wrong media format used in SIP INVITE causes one-way audio

osmo-sip-connector from the latest git repository specifies wrong audio codec in SIP INVITE message.

When using osmo-sip-connector to bridge a nanoBTS and an asterisk server, osmo-sip-connector sends 0 as the media codec which corresponds to G.711 PCMU, causing asterisk to respond in g711 and nanoBTS is unable to parse it.

Attached is a packet capture between the BSC and Asterisk server.

Related issues:
- Related to OsmoMSC - Bug #3650: MSC is not sending a payload_type to MNCC?
- Related to osmo-sip-connector - Bug #5177: respect dynamic RTP payload types ...

Files
- osmo-sip-connector-problem.pcap

History
- #1 - 12/11/2018 05:20 PM - fixeria
  - Related to Bug #3650: MSC is not sending a payload_type to MNCC? added

- #2 - 06/12/2021 07:31 AM - laforge
  - Related to Bug #5177: respect dynamic RTP payload types of external SIP added