Osmo-CC Endpoint Guide

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1. Osmo-CC

1.1 What is Osmo-CC

Osmo-CC is a telephony interface to connect telephone applications and interfaces. Each application or interface is called '**endpoint**'. The applications may run on a local machine or local IP network, or over wide area networks using tunnel or other security features.

It was designed as a replacement for the '**MNCC**' interface that was used by '**openbsc**' software to interconnect with a simple SIP endpoint or '**Linux-Call-Router**' (ISDN and SIP gateway software). The '**MNCC**' interface was re-used for other projects like '**OsmocomBB**' (mobile phone stack) and '**osmocom-analog**' (classic 1G base station emulator). The '**MNCC**' interface was designed for GSM. The messages and state

machine is different for network and mobile side, so that upper layers are not compatible. It does not support overlap dialing, so that it is restricted to be used in the mobile world, where no overlap dialing exists. Information elements are restricted to GSM codecs, hacks were used to add linear audio for '**osmocom-analog**'.

Osmo-CC is made to interface different telephone applications, like ISDN interfaces, POTS interfaces, SIP interfaces, GSM components and more. Instead of using different application parts as in Signaling System No. 7, Osmo-CC provides a union off most messages, call states and information elements for all telephone networks. The application that uses Osmo-CC does not need to consider what network is used. Also it does not need to consider what mode the network protocol uses, like NT or TE, User or Network, FXS or FXO, ect.

1.2 Features of Osmo-CC

The Osmo-CC interface can be expanded with new messages and information elements that may be required by endpoints for new networks. Older existing application will simply ignore these information.

Osmo-CC supports overlap dialing, so that analog phones (FXS) can be used without dial timeout. But it is not required that an endpoint supports overlap sending or receiving or both.

Osmo-CC supports early audio until the call is connected. This way it is possible to hear announcements or tones. Late audio (after the call has been disconnected) is also supported. It is not required that endpoints support sending/receiving early/late audio.

Osmo-CC can be used as an interface inside an application with an endpoint, as well as between endpoints using a TCP/IP socket interface.

1.3 What Osmo-CC does not support

There is no security at Osmo-CC socket interface. This means that no authentication nor encryption is used. This is not a problem, if all applications that use Osmo-CC run on the same machine. In cause connecting application via Osmo-CC over a network, the network operator needs to add some security, like firewall or encrypted tunneling.

Classical telephone networks allow calls to be suspended or resumed at application layer. Osmo-CC does not support any messages for this. Instead, the endpoint that serves a network must handle suspend and resume of calls itself. Still it is possible to notify other endpoints via Osmo-CC messages that the call has been suspended or resumed.

Call forwarding or redirection is not supported by Osmo-CC. Application that require call forwarding must handle it themselves. Similar to changing the destination of a call, changing the audio codec is also not supported.

1.4 Technical detail

Osmo-CC is the interface that connects the network layer of an endpoint to higher layer. Also it can be used to bridge network layers of multiple endpoints via TCP/IP socket without the requirement of any PBX or router.

All messages start with a single byte of message type, followed by two bytes of length (network order) of all information elements, followed by the information elements. Each information element starts with a single byte of information element type, followed by two bytes of length (network order) of the information element, followed by information element data. Information elements may be used as a vector of information element. This means they may be omitted, included once or repeated multiple times within a message.

The interface is almost symmetrical. A release towards lower layer must be confirmed, a release from lower is not responded. (This way the application knows when resources like audio channels are free to be used for new calls.) Also the numbering of caller IDs and dialing can be different in each direction.

A TCP/IP socket, that is implemented in '**libosmocc**' with help of a little processing makes the Osmo-CC interface completely symmetrical. Also '**libosmocc**' implements some '**screening**' lists, to convert caller IDs and dialed numbers. Running two applications on the same machine does not require any IP/port configuration. In this case auto-configuration is used.

1.5 Audio streaming

The actual audio is not streamed via Osmo-CC socket interface. Instead, '**RTP**' is used to transfer audio between two endpoints. This means that each endpoint must be capable of transfer '**RTP**'. Also it must support at least '**a-Law**' and '**mu-Law**' codec. In special setups where both endpoints are designed to use different codec, it is not required to support '**a-Law**' and '**mu-Law**' codecs.

To negotiate codecs between endpoints, '**SDP**' is used. Osmo-CC just forwards '**SDP**' information inside its messages. This means that each endpoint must support '**SDP**' also. The benefit is that codec negotiation and audio transfer is end-to-end, especially when using SIP endpoints. New codecs do not require an update of Osmo-CC. Even video and other media could be used.

2. Endpoints

2.1 osmo-cc-alsa-endpoint

This endpoint can be used to make or receive calls with headset. It can be controlled via console interface. The audio uses '**ALSA**' driver. '**ALSA**' is disabled by default, so that it can be used to make test calls only. Use this as a test utility to test other endpoints.

2.1.1 Installation

```
$ cd ~
$ git clone git://git.osmocom.org/cc/osmo-cc-alsa-endpoint
Cloning into 'osmo-cc-alsa-endpoint'…
```

Change to directory:

```
$ cd osmo-cc-alsa-endpoint
```

Configure and compile:

```
$ sudo apt install gcc # in case you don’t have ‘gcc’ installed
$ sudo apt install make # in case you don’t have ‘make’ installed
$ sudo apt install automake # in case you don’t have ‘automake’ installed
$ sudo apt install libasound2-dev # in case you don’t have ALSA development library installed
$ autoreconf –if
$ ./configure
$ make clean
$ make
```

Install on your system:

```
$ sudo make install
```

2.1.2 Example: Call between two ALSA endpoints

Open a first terminal and run ‘osmo-cc-alsa-endpoint’ with given caller ID ‘0815’:

```
$ osmo-cc-alsa-endpoint -I 0815
…
on hook: ............................... (press digits 0..9 or d=dial)
```

Then open a second terminal on the same machine and run ‘osmo-cc-alsa-endpoint’ with some other caller ID or no caller ID:

```
$ osmo-cc-alsa-endpoint
…
on hook: ............................... (press digits 0..9 or d=dial)
```

Now enter some phone number on the first terminal:

```
on hook: 123............................ (press digits 0..9 or d=dial)
```

Then press ‘d’ to dial:

```
connected: 123 (press h=hangup)
```

Now you see the incoming call on the first terminal:

```
connected: 0815->123 (press h=hangup)
```

Press ‘h’ to hangup on either side, to return into ‘on hook’ state.

2.1.3 Using a Headset

To actually use sound, you need to find out the ALSA device of your Headset. To find out, use ‘arecord –l', which shows all sound devices that have input capability:

```
$ arecord -l
**** List of CAPTURE Hardware Devices ****
card 1: Generic [HD-Audio Generic], device 0: ALC887-VD Analog [ALC887-VD Analog]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
card 1: Generic [HD-Audio Generic], device 2: ALC887-VD Alt Analog [ALC887-VD Alt Analog]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
```

The headset on my machine is connected to the first entry in the list above. It is connected to ‘card 1’, device ‘0’. This means that the ALSA device name is ‘hw:1,0’, so I add this option:

```
$ osmo-cc-alsa-endpoint –I 0815 -a hw:1,0
```

2.2 osmo-cc-sip-endpoint

This endpoint can be used to transfer calls via SIP protocol. It can be used without registering/authentication (peer-to-peer), but also with registering/authentication in two directions. This way it is possible to register to a SIP proxy, as well as being a SIP proxy that some device registers to.

The endpoint requires Sofia-SIP stack to be installed. It can be downloaded from http://sofia-sip.sourceforge.net/download.html or alternatively from http://download.eversberg.eu/sofia/sofia-sip-1.12.11.tar.gz.

Note: Compiling with regular optimization causes corrupt SIP messages.

2.2.1 Installation

Installation of Sofia-Sip:

```
$ cd ~
$ wget http://download.eversberg.eu/sofia/sofia-sip-1.12.11.tar.gz
```

Unpack and change to directory:

```
$ tar xvf sofia-sip-1.12.11.tar.gz
$ cd sofia-sip-1.12.11/
```

Compile without optimization:

```
$ sudo apt install gcc # in case you don’t have ‘gcc’ installed
$ sudo apt install make # in case you don’t have ‘make’ installed
$ ./configure CFLAGS=-O0
$ make clean
$ make
```

Install on your system. Be sure that you remove other versions of Sofia-Sip.

```
$ sudo make install
$ sudo ldconfig
```

Installation of osmo-cc-sip-endpoint:

```
$ cd ~
$ git clone git://git.osmocom.org/cc/osmo-cc-sip-endpoint
Cloning into 'osmo-cc-sip-endpoint'...
```

Change to directory:

```
$ cd osmo-cc-sip-endpoint
```

Configure and compile:

```
$ cd ~
$ git clone git://git.osmocom.org/cc/osmo-cc-alsa-endpoint
Cloning into 'osmo-cc-alsa-endpoint'…
```

$ sudo apt install automake # in case you don’t have ‘automake’ installed
$ autoreconf –if
$ ./configure
$ make clean
$ make

Install on your system:

```
$ sudo make install
```

2.2.2 Wardialing

Before running SIP endpoint that is reachable from everywhere on the Internet, note that there are many ‘bots’ that scan the Internet and try to find bad configured SIP devices. Once found, they can be abused to make expensive long-distant calls. If you don’t plan to link your SIP endpoint to the public telephone network, the ‘bots’ might still be annoying and make your phone (or whatever is linked to your SIP endpoint) ring all the time.

Look at this example, where my router receives SIP messages from time to time. (at least the upper three)

```
$ tcpdump -n -i ppp0 udp port 5060
tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on ppp0, link-type LINUX_SLL (Linux cooked), capture size 262144 bytes
08:19:28.226120 IP 167.114.117.174.5086 > 82.139.198.227.5060: SIP, length: 419
08:25:29.389918 IP 193.107.216.17.5189 > 82.139.198.227.5060: SIP, length: 415
09:42:49.046676 IP 45.134.144.4.5071 > 82.139.198.227.5060: SIP, length: 405
09:59:50.361642 IP 101.37.32.76.26242 > 82.139.198.227.5060: SIP, length: 16
```

If you think that authentication prevents ‘bots’ from making calls, you are wrong. If you register and make calls with authentication to some SIP proxy, you are not safe. If there is an incoming call from that proxy, no authentication is used in this direction. You need to prevent ‘bots’ from reaching your SIP endpoint. The best way is to use a firewall to block all incoming SIP messages, except for messages from the proxy.

By default, local SIP port is 5060. If you run multiple SIP endpoints on one machine, you need other port numbers. Changing the port number instead of blocking IPs is not enough. I have seen ‘bots’ that scan other ports also.

If you use NAT, you are safe, because NAT will only forward packets of connections that have been previously initiated by the SIP endpoint.

If you use port forwarding, limit that to the IP of the proxy or any other remote SIP peer you want to connect with.

If you have direct connection without NAT and without port forwarding, limit incoming packets to established connections. Configuring your firewall is beyond the scope of this document.

2.2.3 Example: peer-to-peer

For better debugging SIP messages, use ‘**sngrep**’. It will show all SIP sessions that run via local machine. Install it on your machine with SIP endpoint or on your router, if it runs on linux. After start it will capture all SIP packets. The first screen displays all session. Use the cursor and enter to select a session and switch to the second screen. The seconds screen shows a session with all messages. Use the cursor and enter to select a message and switch to the third screen. On the third screen the message is shown. Use Escape to go back one screen and to end program.

Run osmo-cc-alsa-endpoint on machine 1. The machine has local IP address ‘10.0.0.28’, so we need to tell the RTP process our local IP. We set our caller ID to ‘0815’. We also add a phone number, so that we don’t need to enter after starting the endpoint.

```
$ ‘osmo-cc-alsa-endpoint’ -a default --cc "rtp-peer 10.0.0.28" -I 0815 123
...
options.c: 282 info   : Command line option '--cc', parameter 'rtp-peer 10.0.0.28'
options.c: 268 info   : Command line option '-I' ('--caller-id'), parameter '0815'
socket.c: 381 error  : OsmoCC-Socket failed, socket cause 3.
endpoint.c: 923 info   : Handle message CC-REL-REQ at state ATTACH-SENT (callref 1)
endpoint.c: 314 info   : Attachment to remote peer "127.0.0.1:4201" failed, retrying.
...
```

Also run ‘osmo-cc-sip-endpoint’ in different terminal on machine 1. The remote machine has IP address ‘10.0.0.157’. We tell SIP endpoint our local and the remote IP.

```
$ osmo-cc-sip-endpoint -l 10.0.0.28 -r 10.0.0.157
...
options.c: 268 info   : Command line option '-l' ('--local'), parameter '10.0.0.28'
options.c: 268 info   : Command line option '-r' ('--remote'), parameter '10.0.0.157'
endpoint.c: 923 info   : Handle message CC-ATTACH-RSP at state ATTACH-SENT (callref 1)
endpoint.c: 299 info   : Attached to remote peer "127.0.0.1:4200".
endpoint.c: 923 info   : Handle message CC-ATTACH-REQ at state IDLE (callref 2)
endpoint.c:  381 info     :  Remote  peer  with  socket  address  '127.0.0.1'  and  port  '4200'  and
interface 'alsa' attached to us.
endpoint.c: 385 info   : Changing message to CC-ATTACH-CNF.
...
```

On machine 2, run ‘omso-cc-alsa-endpoint’. Note that our local machine has local IP ‘10.0.0.157’.

```
$ ‘osmo-cc-alsa-endpoint’ -a default --cc "rtp-peer 10.0.0.157"
...
options.c: 282 info   : Command line option '--cc', parameter 'rtp-peer 10.0.0.157'
options.c: 268 info   : Command line option '-I' ('--caller-id'), parameter '0815'
socket.c: 381 error  : OsmoCC-Socket failed, socket cause 3.
endpoint.c: 923 info   : Handle message CC-REL-REQ at state ATTACH-SENT (callref 1)
endpoint.c: 314 info   : Attachment to remote peer "127.0.0.1:4201" failed, retrying.
...
```

Also run ‘osmo-cc-sip-endpoint’ in different terminal on machine 2. Note that the remote machine has IP address ‘10.0.0.28’.

```
$ osmo-cc-sip-endpoint -l 10.0.0.157 -r 10.0.0.28
...
options.c: 268 info   : Command line option '-l' ('--local'), parameter '10.0.0.157'
options.c: 268 info   : Command line option '-r' ('--remote'), parameter '10.0.0.28'
endpoint.c: 923 info   : Handle message CC-ATTACH-RSP at state ATTACH-SENT (callref 1)
endpoint.c: 299 info   : Attached to remote peer "127.0.0.1:4200".
endpoint.c: 923 info   : Handle message CC-ATTACH-REQ at state IDLE (callref 2)
endpoint.c:  381 info     :  Remote  peer  with  socket  address  '127.0.0.1'  and  port  '4200'  and
interface 'alsa' attached to us.
endpoint.c: 385 info   : Changing message to CC-ATTACH-CNF.
...
```

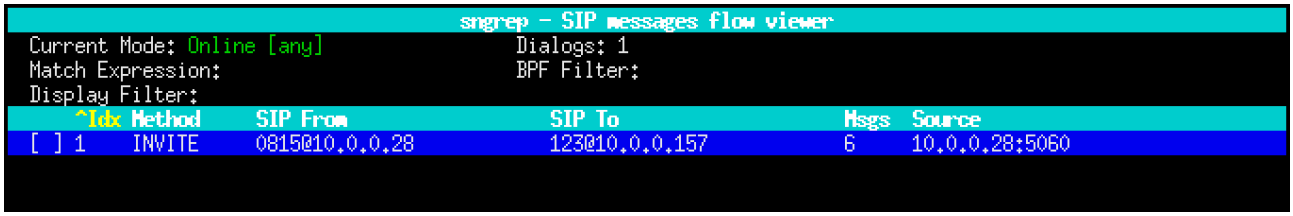
To debug a SIP call, run ‘sngrep’ on one of the machines in another terminal. Now press ‘d’ to dial the on machine 1 at console of ‘osmo-alsa-endpoint’.

```
telephone.c: 642 info   : Outgoing call from '0815' to '123'
endpoint.c: 923 info   : Handle message CC-SETUP-IND at state IDLE (callref 361)
endpoint.c: 923 info   : Handle message CC-PROC-REQ at state INIT-IN (callref 361)
telephone.c: 484 info   : Incoming call acknowledged
endpoint.c: 923 info   : Handle message CC-SETUP-RSP at state PROCEEDING-IN (callref 361)
telephone.c: 503 info   : Incoming call acknowledged
endpoint.c: 923 info   : Handle message CC-SETUP-COMP-IND at state CONNECTING-IN (callref 361)
connected: 123 (press h=hangup)
```

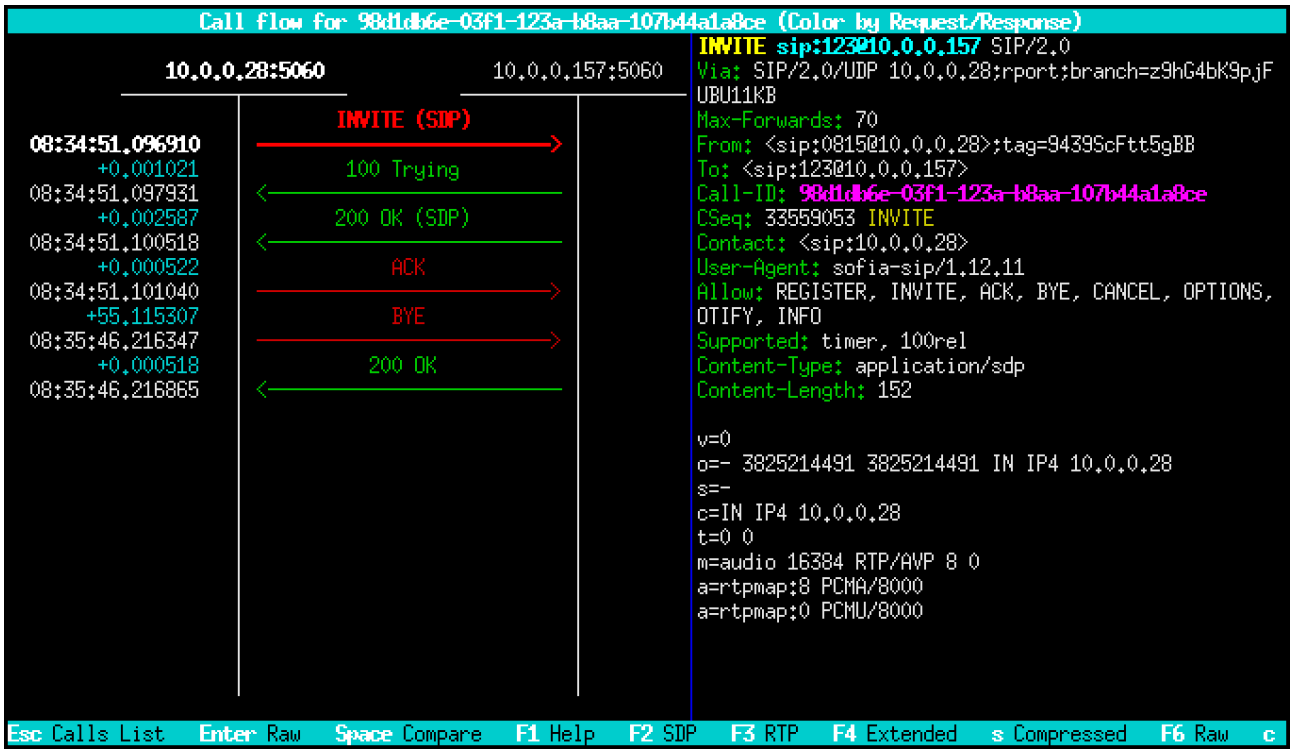
Then press ‘h’ to end the call.

```
telephone.c: 686 info   : Call hangup
endpoint.c: 923 info   : Handle message CC-REL-IND at state ACTIVE (callref 361)
endpoint.c: 740 info   : Changing message to CC-DISC-IND.
endpoint.c: 923 info   : Handle message CC-REL-REQ at state DISCONNECTING-IN (callref 361)
on hook: 123.............................. (press digits 0..9 or d=dial)
```

Watch the debug output on all four terminals. Also look at the ‘sngrep’ terminal:



Press enter to see a detailed graph of the call you just made:



You can see in the ‘INVITE’ message your local and remote IP, as well as local and remote phone number. The SDP body attached to the ‘INVITE’ message contains the supported codecs (PCMU and PCMA), the RTP IP ‘10.0.0.28’ and port ‘16384’ of machine 1. Use cursor keys to view the other messages.

TBD

2.2.4 Example: Register to a SIP proxy

TBD

2.2.5 Example: Configure as SIP proxy

TBD

2.2.6 Options

--send-ner

When a call is received, the SIP endpoint may send audio prior answer of the call. In case that audio is available pior answer, response to incoming call is “183 Session Progress” prior ringing (“180 Ringing”). If the remote has trouble with extra ringing message, disable it with “--send-ner”.

--receive-ner

When an outgoing SIP call is made, the remote SIP endpoint may respond with “183 Session Progress”, to indicate that audio is available prior answer. If it does not respond with ringing afterwards (“180 Ringing”), use “--receive-ner” to treat “183 Session Progress” as if the call is ringing.

--asserted-id user@host

If you register to a proxy, you might have a user name and a host name to register to. Then if you make calls, you might want to replace your source user name with your caller ID. This is useful if you have the ability to give any caller ID or if you have a set of phone number or extensions. The proxy will not know who you are, so it cannot associate your call with your identity. To solve this, use “--asserted-id”.

--public-ip <ip>

If you your SIP endpoint is located behind NAT firewall and your remote SIP peer is outside your network, you can specify the public IP that your firewall will translate local IP addresses to. If you don’t know your IP, use “--stun-server” instead.

This will not only add your contact with public IP to SIP messages, but also change contact inside SDP message. Then the remote knows where it can reach you and where to send audio data to.

--stun-server <server ip>

If you are behind a NAT firewall and have dynamic IP, use a stun server to obtain public IP address. SIP and SDP messages then use this IP as public IP address.

--register-interval <seconds>

Define interval for registering. If you experience that you unreachable between two register intervals, use shorter value. Lower value will also prevent NAT firewall from aging out.

--options-interval <seconds>

Same as above, but for option messages during a call.

--stun-interval <seconds>

Interval to ask STUN server for your current public IP.

--expires <seconds>

Set session expire timer. This will cause a re-invite. Turn off using ‘0’, which is the default value.

2.3 osmo-cc-isdn-endpoint

TBD
- verweis Siehe anhang
- nt or te
- ptp or ptmp
- ptmp->msn
- ptp->l1 / l2 hold
- fax? –tx-delay

2.4 osmo-cc-ss5-endpoint

TBD
kurzinfo

2.5 Osmocom-Analog

TBD
- don’t use -a for console
- use -o to use socket
- use -x to connect to itself (cross connet calls)

-Example zeitansage
-Example: anruf per alsa.

3. Routing & Screening

- dialing screening
- both directions
- caller id screening
- both directions, connected id
- routing (local/remote)
- routing via screening

4. osmo-cc-router

TBD
- script erklaeren, wann und wie und bash
- script endet, neustart beim overlap
- script endet nach call, call geht weiter
- echo stdout
- echo stderr

commands:
- rtp-proxy (DTMF/transcoding/call-recording/announcements)
 what is default
 orig-codecs
 term-codecs
  PCMA PCMU L16 telephone-event

- play <name> (rtp-proxy)
  volume (factor)
  loop
- play-stop
- record <name> (rtp-proxy)
  volume (factor)
- record-stop
- tx-gain <db>, rx-gain <db>
- call
                if (value_of_param(argv[i], "interface", &interface));
            else if (value_of_param(argv[i], "bearer-coding", &bearer_coding));
                                else if (value_of_param(argv[i], "bearer-capability", &bearer_capability));
            else if (value_of_param(argv[i], "bearer-mode", &bearer_mode));
            else if (value_of_param(argv[i], "calling", &calling));
            else if (value_of_param(argv[i], "calling-type", &calling_type));
            else if (value_of_param(argv[i], "calling-plan", &calling_plan));
            else if (value_of_param(argv[i], "calling-present", &calling_present));
            else if (value_of_param(argv[i], "calling-screen", &calling_screen));
            else if (value_of_param(argv[i], "no-calling", &no_calling));
            else if (value_of_param(argv[i], "calling2", &calling2));
            else if (value_of_param(argv[i], "calling2-type", &calling2_type));
            else if (value_of_param(argv[i], "calling2-plan", &calling2_plan));
                                else if (value_of_param(argv[i], "calling2-present", &calling2_present));
                                 else if (value_of_param(argv[i], "calling2-screen", &calling2_screen));
            else if (value_of_param(argv[i], "no-calling2", &no_calling2));
            else if (value_of_param(argv[i], "redirecting", &redirecting));
                                else if (value_of_param(argv[i], "redirecting-type", &redirecting_type));
                                else if (value_of_param(argv[i], "redirecting-plan", &redirecting_plan));
                            else if (value_of_param(argv[i], "redirecting-present", &redirecting_present));

else if (value_of_param(argv[i], "redirecting-screen", &redirecting_screen));
else if (value_of_param(argv[i], "redirecting-reason", &redirecting_reason));
else if (value_of_param(argv[i], "no-redirecting", &no_redirecting));
else if (value_of_param(argv[i], "dialing", &dialing));
else if (value_of_param(argv[i], "dialing-type", &dialing_type));
else if (value_of_param(argv[i], "dialing-plan", &dialing_plan));
else if (value_of_param(argv[i], "keypad", &keypad));

- call-stop
- overlap, proceeding, alerting, answer
- disconenct, release
  cause
- dtmf
  → stdout: dtmf <digit>
- dtmf stop
- error
  → debug message

telephone-event and dtmf:
 dtmf x

Appendix

A. ISDN-Hardware

TBD