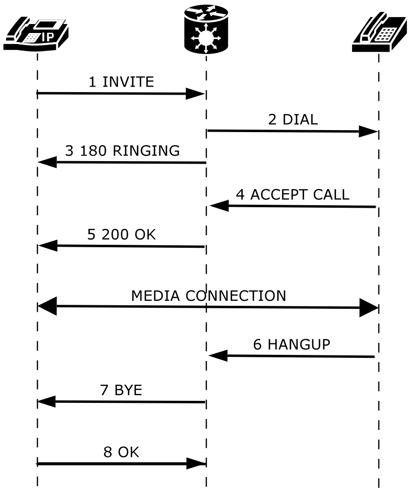
isdngw implementation status

12. Juli 2003

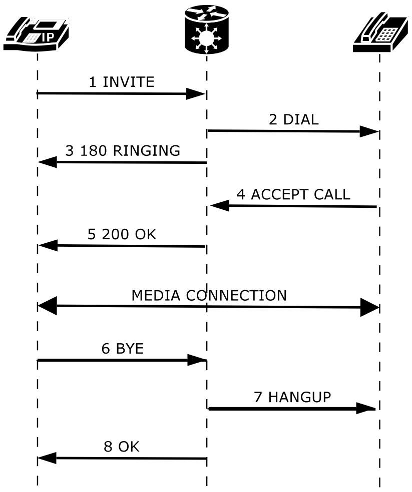
1 call flow implementation status of sems/isdngw

	description	implemented	fully tested
1.1	SIP2PSTN call ended by PSTN	yes	no
1.2	SIP2PSTN call ended by SIP UA	yes	no
1.3	SIP2PSTN call with busy PSTN line	yes	no
1.4	SIP2PSTN call unanswered in PSTN	yes	no
1.5	SIP2PSTN cancel by SIP before PSTN accept	no	no
1.6	PSTN2SIP call ended by PSTN hangup	partly	no
1.7	PSTN2SIP call ended by SIP UA	yes	no
1.8	PSTN2SIP SIP call setup failed	yes	no
1.9	PSTN2SIP canceled by PSTN before SIP accept	party	no

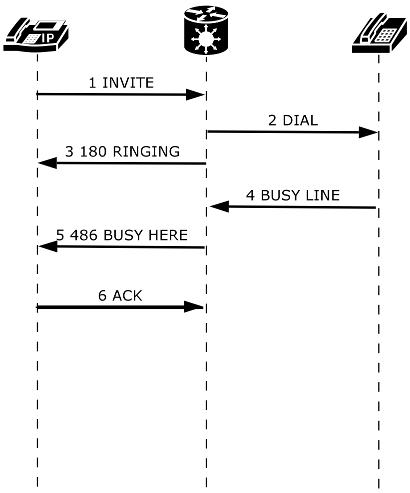
1.1 SIP2PSTN call ended by PSTN



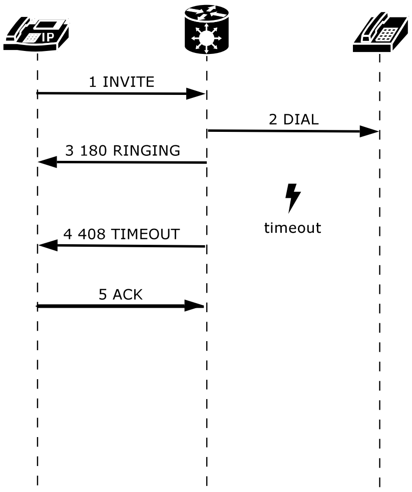
1.2 SIP2PSTN call ended by SIP UA



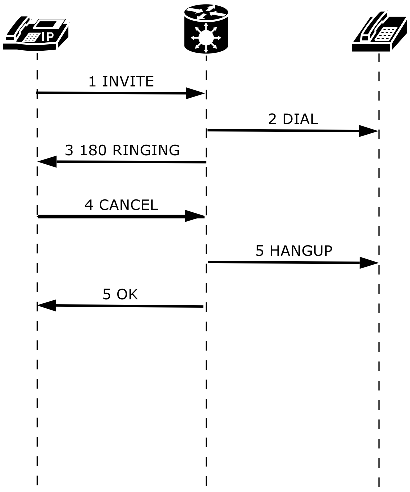
1.3 SIP2PSTN call with busy PSTN line



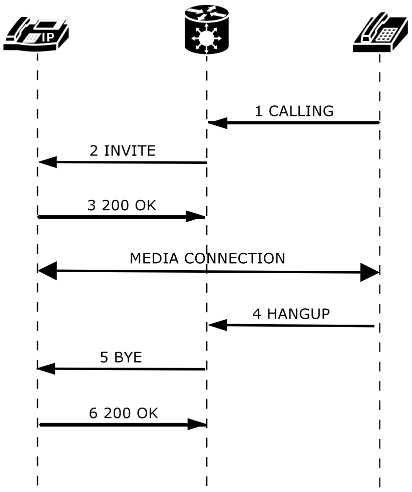
1.4 SIP2PSTN call unanswered in PSTN



1.5 SIP2PSTN cancel by SIP before PSTN accept

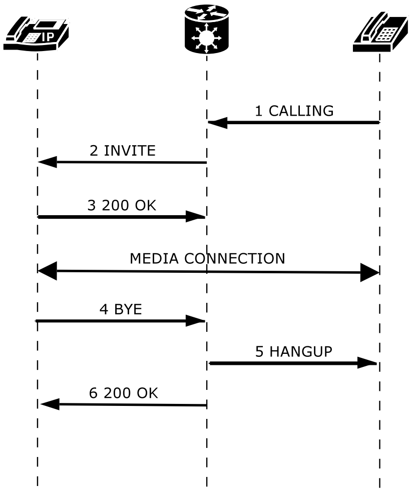


1.6 PSTN2SIP call ended by PSTN hangup

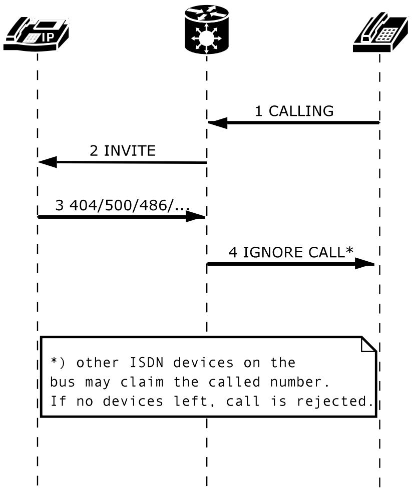


TODO: check why kphone discards the BYE... do some testing with other UAs.

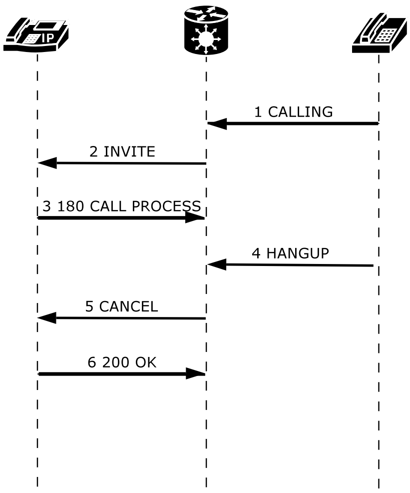
1.7 PSTN2SIP call ended by SIP UA



1.8 PSTN2SIP SIP call setup failed



1.9 PSTN2SIP canceled by PSTN before SIP accept



TODO: track down 481 from ser after call is canceled already...

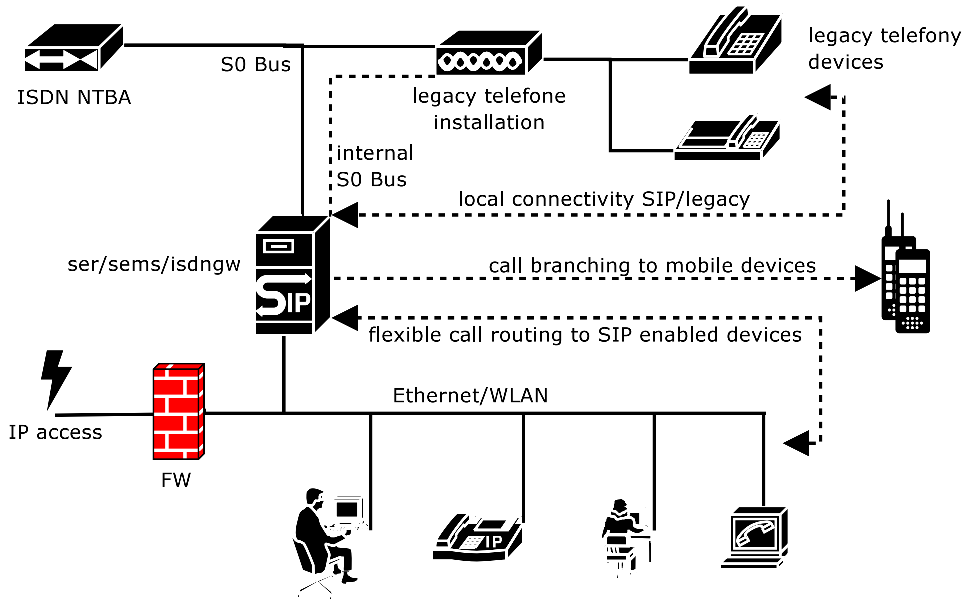
2 drawbacks using isdn4linux' ttyI devices

Using the default linux isdn api has the big advantage, that the gateway works with any linux-supported voice capable isdn card (even the passive ones) including professionell cards. But there are some issues, that cannot be solved due to API insufficiencies:

- There is no way to detect the RINGING state, the current implementation assumes ringing after a timeout of 3 seconds.
- CLIR is not available
- PSTN callers always hears a ringing, even if no SIP UAS is ringing.
- There is no way to signal BUSY (besides if the gateway itself has no b-channel available), a busy on SIP side results as any other error in a rejected call.

3 small office scenario

The current implementation allows the following scenario, which already offers some useful functions for small offices. It assumes a small office with networked SIP-enabled workstations, a legacy telephony system and a server running ser/sems/isdngw.



The SIP server interoperates with legacy telephone systems and offers additional features like turning workstations into full featured telephones and call forking to mobile phones. Incoming calls can be signalized on legacy devices and any number of SIP devices at the same time, if any device is answered, the rest stops signalling.

All this can be deployed without touching the configuration of the telephone system.