

Příloha A

Modifikace zdrojového kódu PBX Asterisk v souboru *Asterisk/channels/chan_sip.c* dle [23]. Úprava začíná na řádku 14079 a končí na řádku 14092.

```
/* Find out what they require */
required = get_header(req, "Require");
if (!ast_strlen_zero(required)) {
    required_profile = parse_sip_options(NULL,
        required);
    if (required_profile && required_profile !=
        SIP_OPT_REPLACES) {
        /* At this point we only support REPLACES */
// transmit_response_with_unsupported(p, "420
Bad extension (unsupported)", req, required);
        ast_log(LOG_WARNING, "Received SIP INVITE
            with unsupported required extension: %s\
            n", required);
        p->invitestate = INV_COMPLETED;
        if (!p->lastinvite)
            sip_scheddestroy(p, DEFAULT_TRANS_TIMEOUT);
// return -1;
    }
}
```