La Plateforme **Asterisk – From Scratch**

On installe une VM avec Debian 12 en mode console (cocher SSH et outils systèmes seulement)

on autorise ssh sur le compte root :
echo "PermitRootLogin Yes" > /etc/ssh/sshd_config.d/ssh.conf

On installe les paquets nécessaires pour compiler Asterisk :

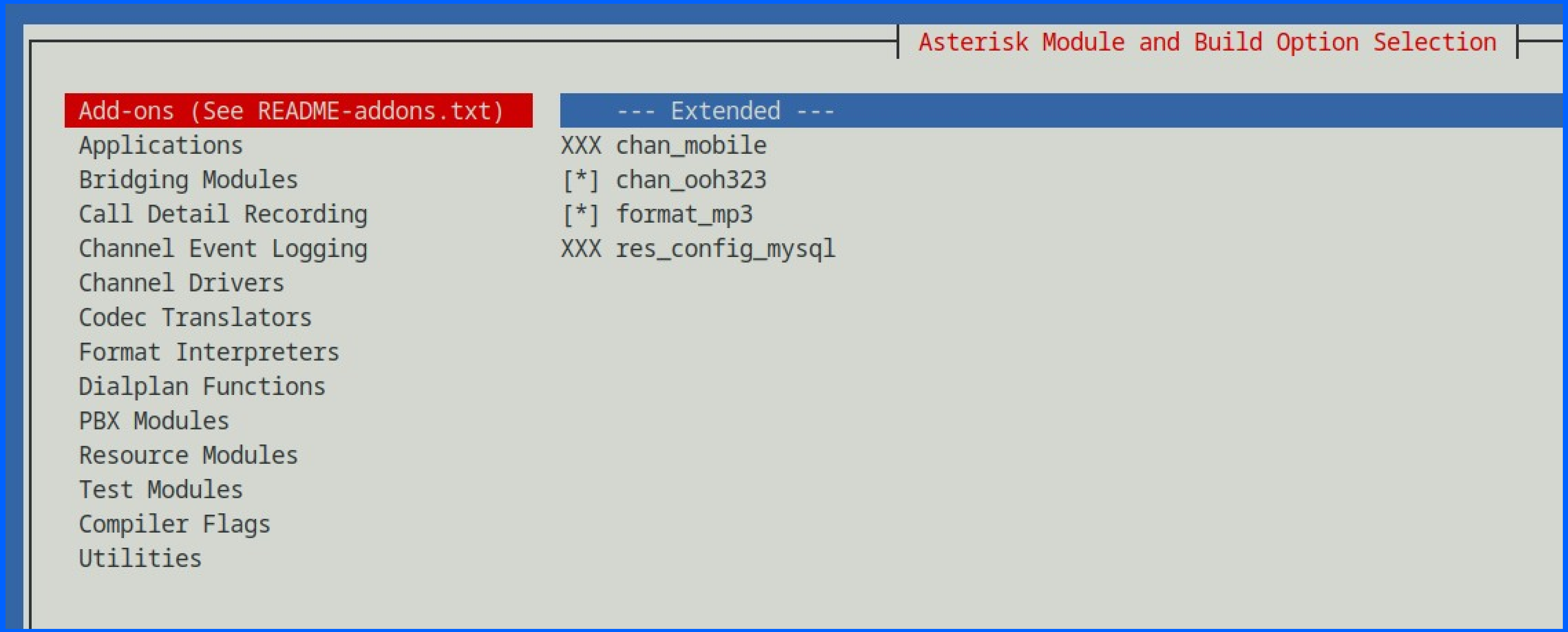
apt install git curl build-essential libedit-dev git curl wget libnewt-dev libssl-dev libncurses5-dev subversion libsqlite3-dev build-essential libjansson-dev libxml2-dev uuid-dev avahi-daemon

On Télécharge la dernière version d'Asterisk :

```
cd /usr/src
apt install wget
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-22-current.tar.gz
tar xfv asterisk-22-current.tar.gz
rm asterisk-22-current.tar.gz
cd asterisk-22*

./configure
contrib/scripts/get_mp3_source.sh
```

On configure les options par : make menuselect



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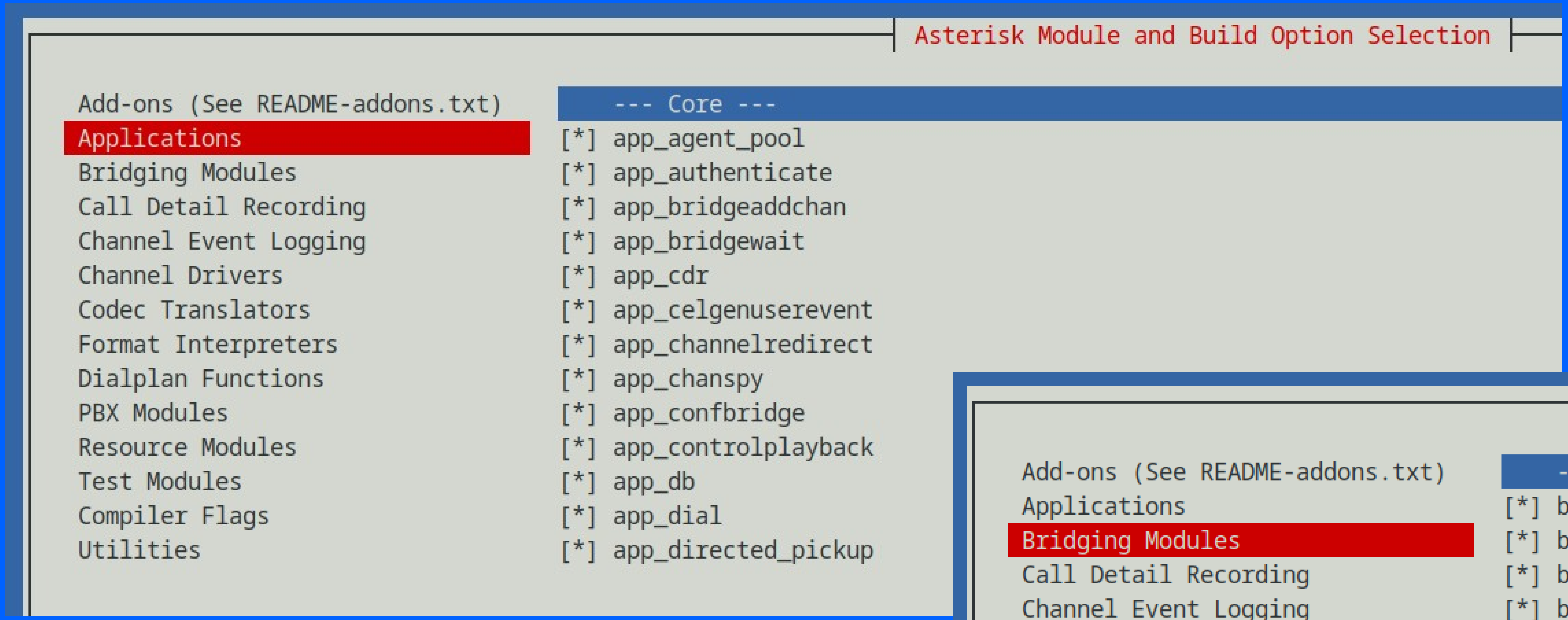

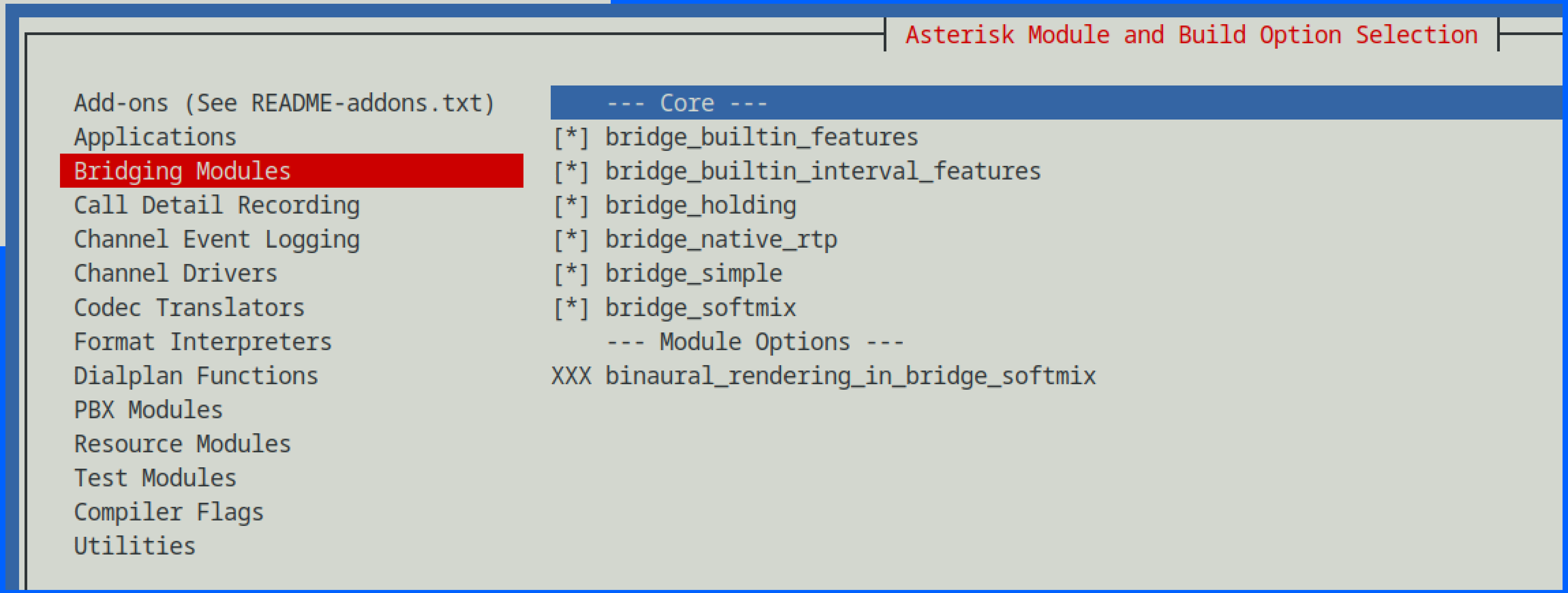

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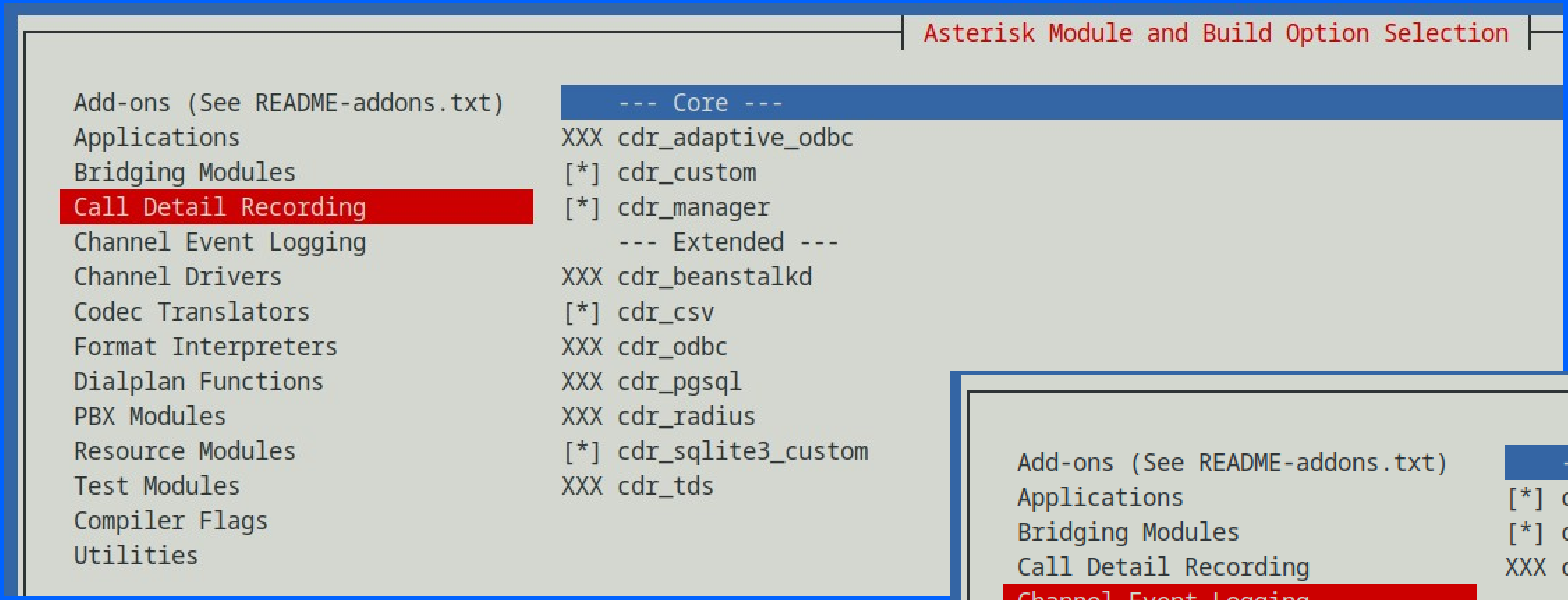

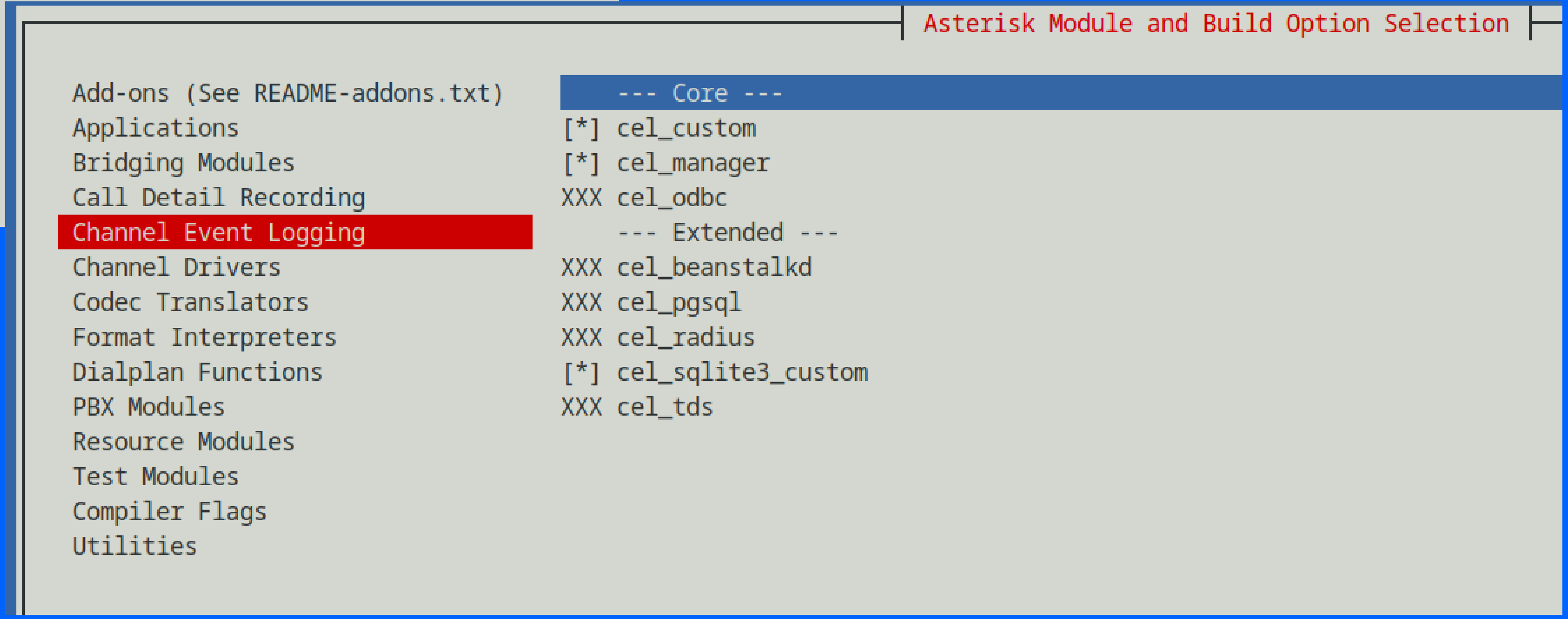

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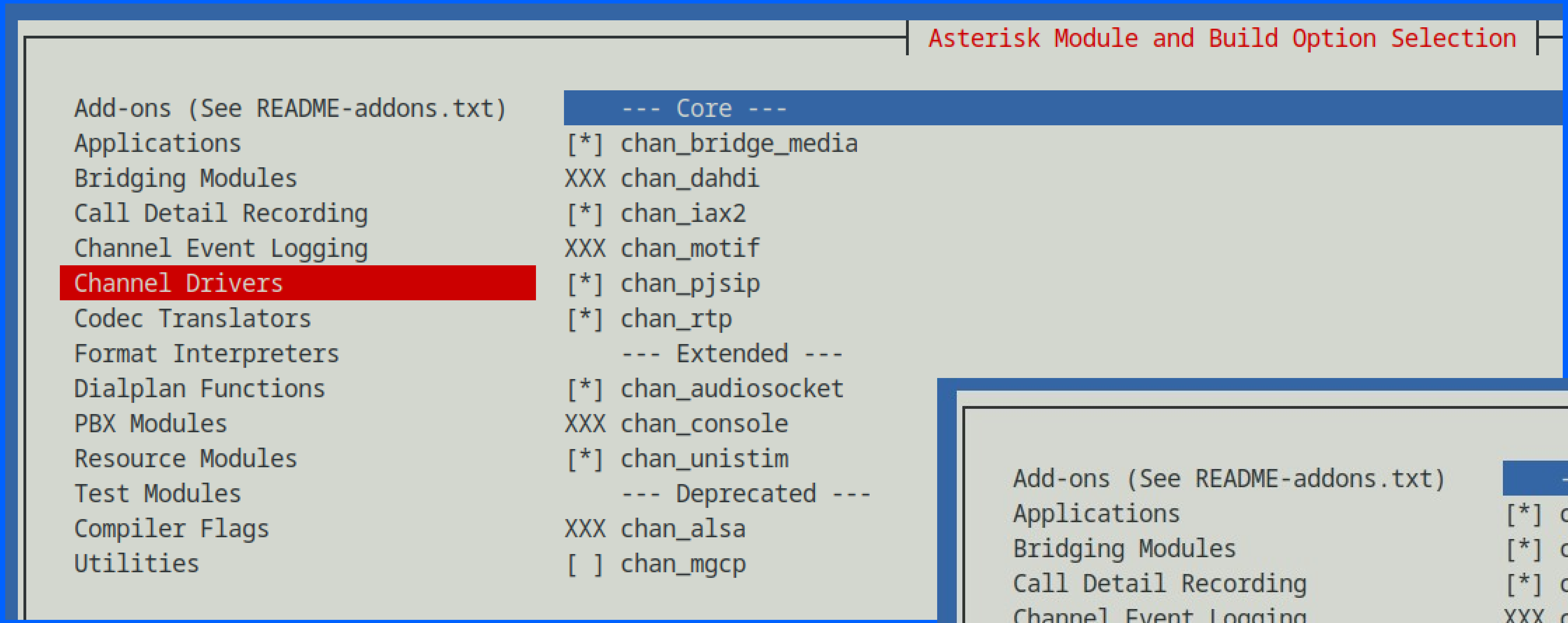
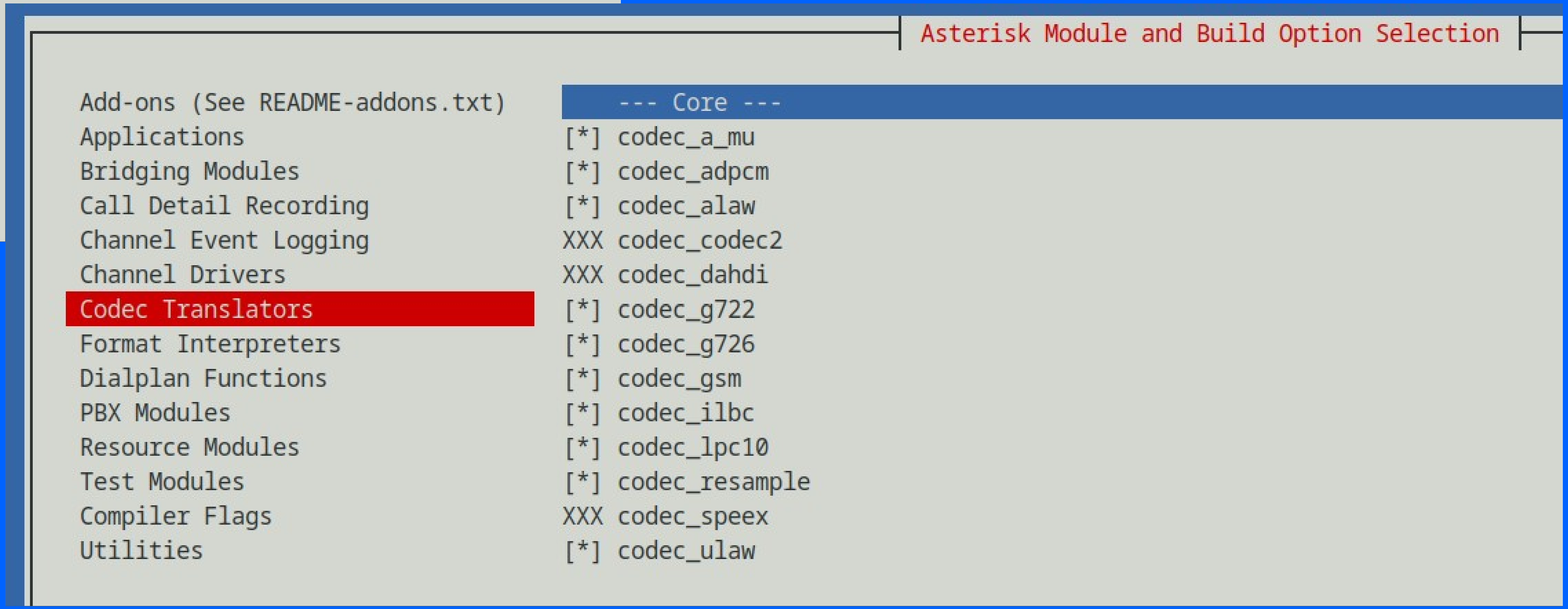




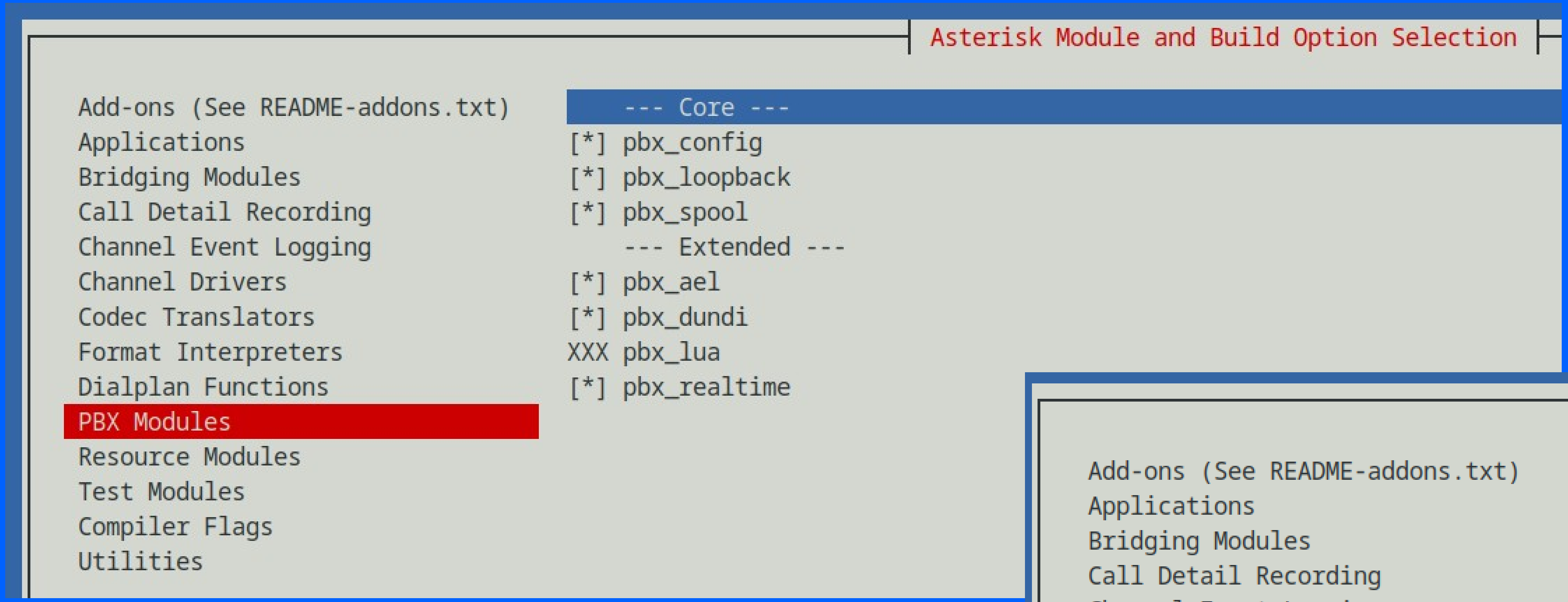
La Plateforme Asterisk – From Scratch

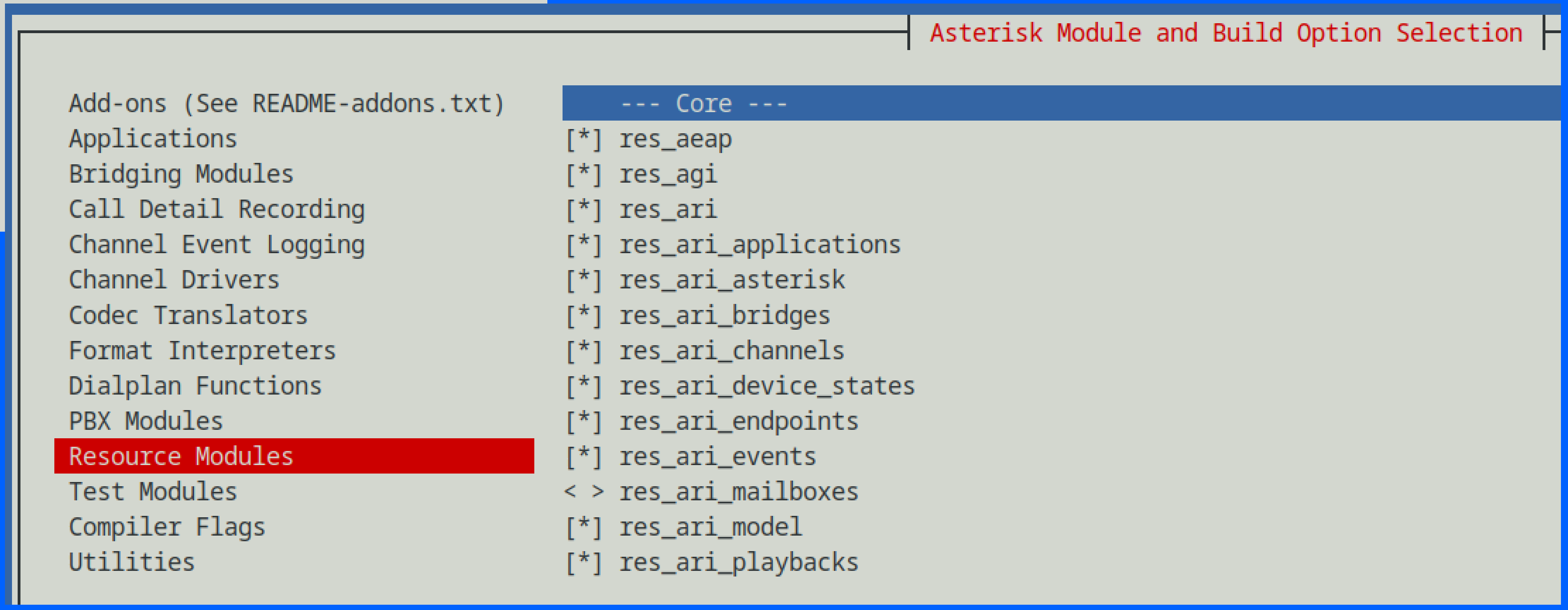
```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] format_g719
Bridging Modules                    [*] format_g723
Call Detail Recording               [*] format_g726
Channel Event Logging               [*] format_g729
Channel Drivers                     [*] format_gsm
Codec Translators                   [*] format_h263
Format Interpreters                 [*] format_h264
Dialplan Functions                  [*] format_ilbc
PBX Modules                         XXX format_ogg_vorbis
Resource Modules                    [*] format_pcm
Test Modules                        [*] format_siren14
Compiler Flags                      [*] format_siren7
Utilities                           [*] format_sln
```

```
Asterisk Module and Build Option Selection

Add-ons (See README-addons.txt)     --- Core ---
Applications                        [*] func_aes
Bridging Modules                    [*] func_base64
Call Detail Recording               [*] func_blacklist
Channel Event Logging               [*] func_callcompletion
Channel Drivers                     [*] func_callerid
Codec Translators                   [*] func_cdr
Format Interpreters                 [*] func_channel
Dialplan Functions                  [*] func_config
PBX Modules                         XXX func_curl
Resource Modules                    [*] func_cut
Test Modules                        [*] func_db
Compiler Flags                      [*] func_devstate
Utilities                           [*] func_dialgroup
```

La Plateforme

Asterisk – From Scratch





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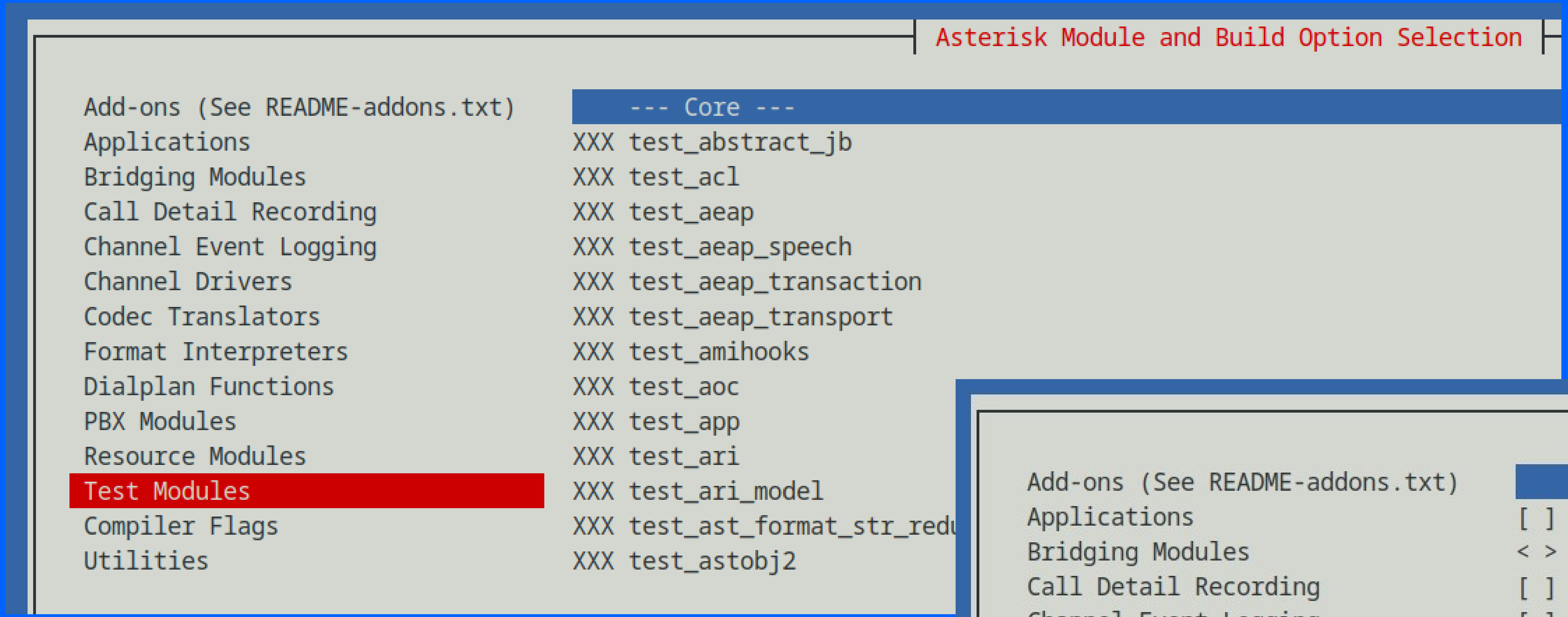

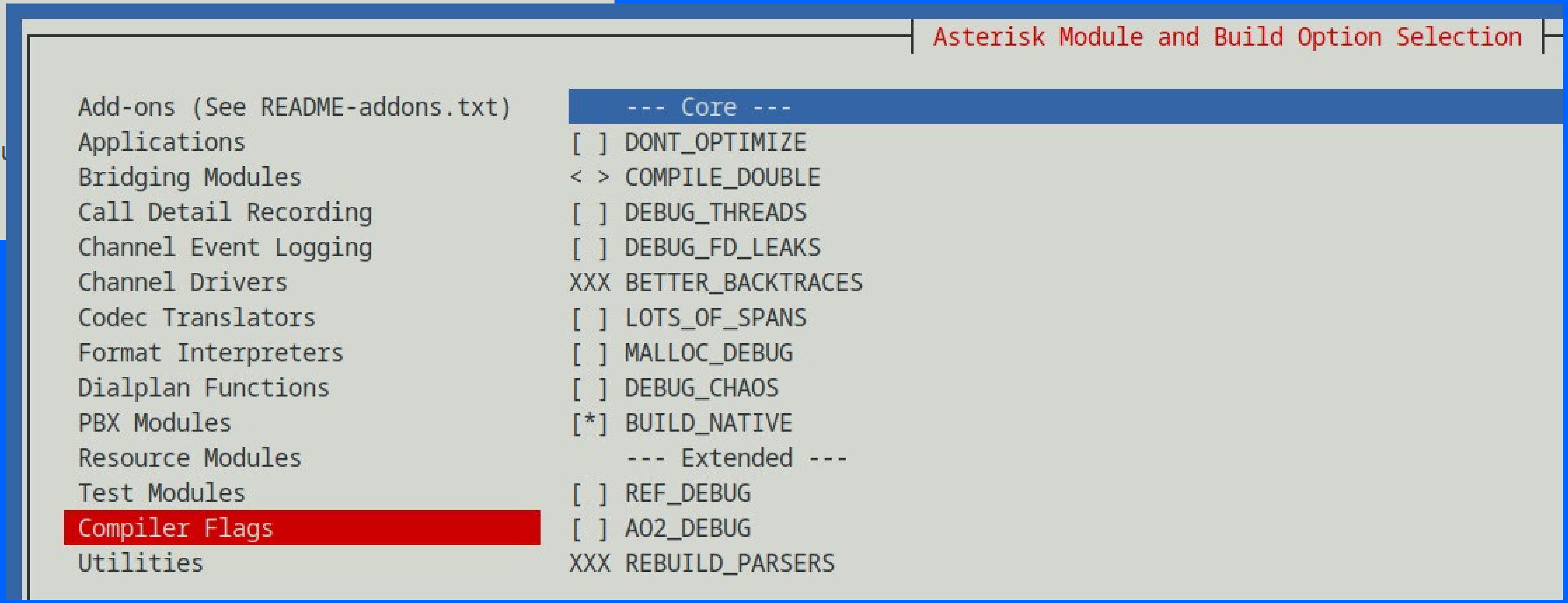

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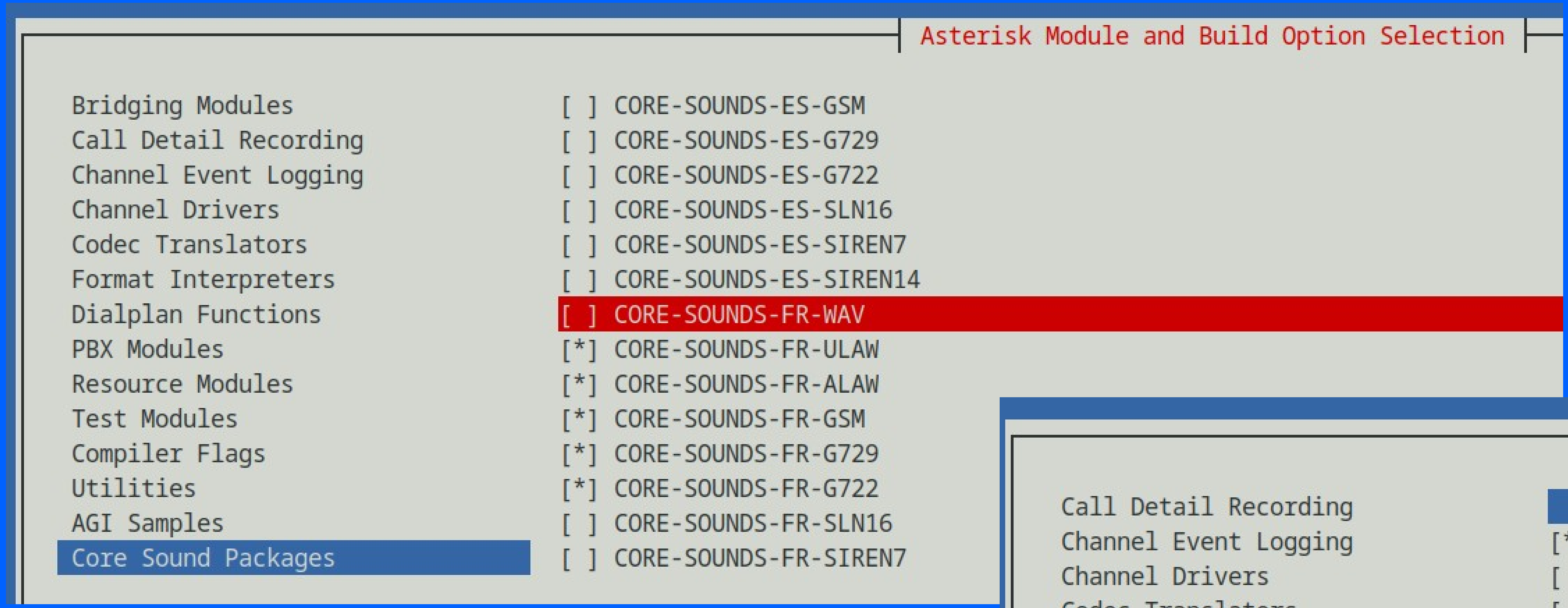
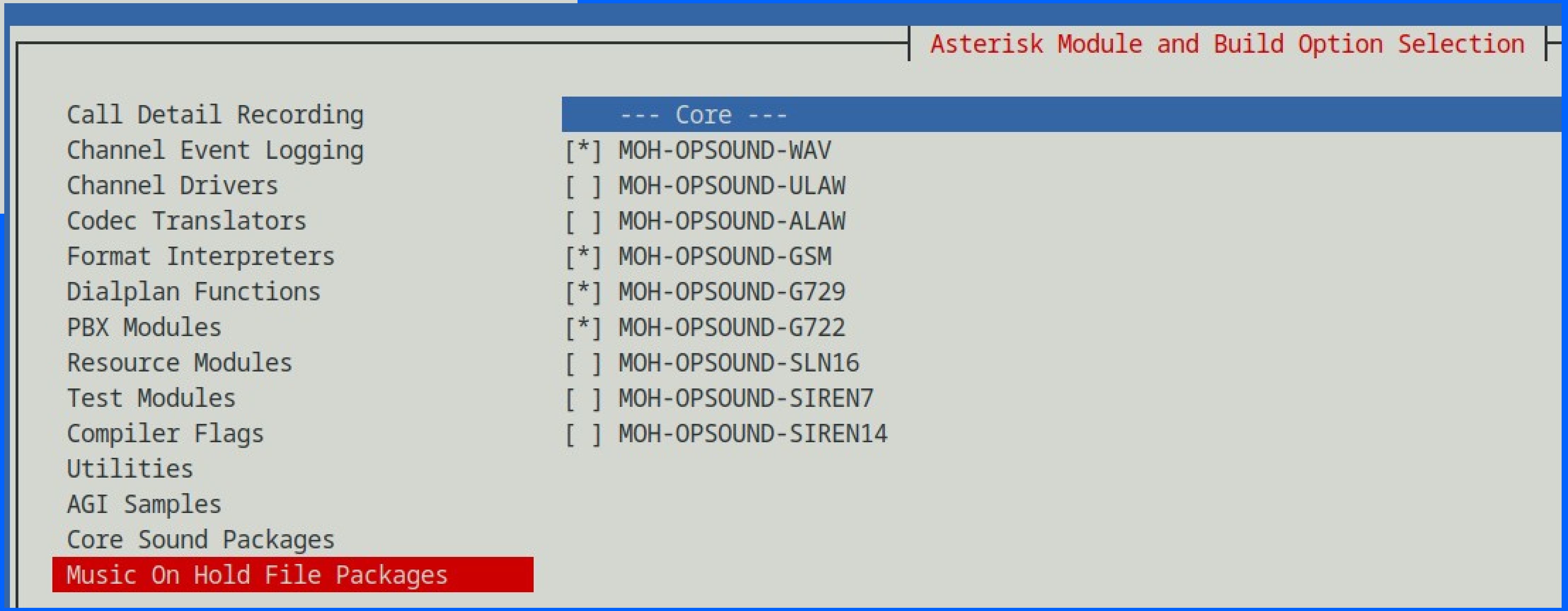
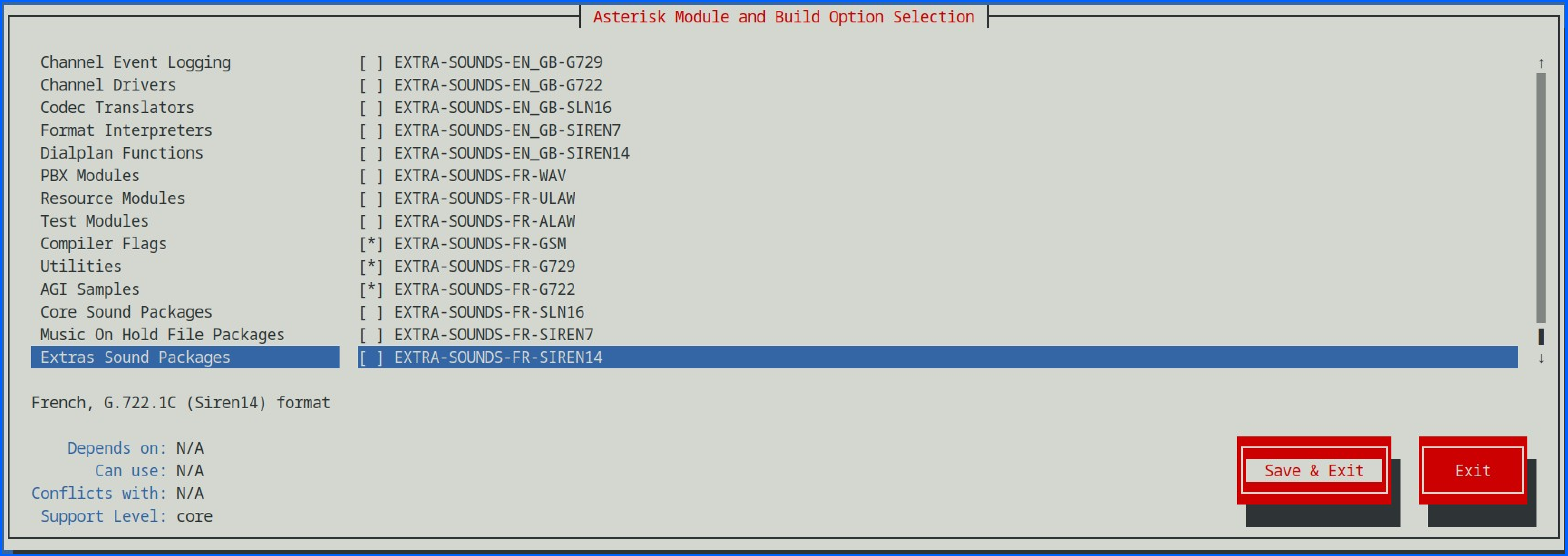




Asterisk – From Scratch



Une fois la configuration terminée on compile :

make
make install
make samples
make basic-pbx
make config

Selon la puissance de la machine , l'opération peux être longue .

Asterisk – From Scratch

Une fois terminé on lance Asterisk et on vérifie son status :

service asterisk start

service asterisk status

```
root@pbx:/usr/src/asterisk-20.3.0# service asterisk status
● asterisk.service - LSB: Asterisk PBX
     Loaded: loaded (/etc/init.d/asterisk; generated)
     Active: active (running) since Tue 2023-06-20 08:31:46 CEST; 2h 9min ago
       Docs: man:systemd-sysv-generator(8)
      Tasks: 39 (limit: 2265)
     Memory: 34.7M
        CPU: 1min 28.284s
     CGroup: /system.slice/asterisk.service
             └─2373 /usr/sbin/asterisk

juin 20 08:31:46 pbx systemd[1]: Starting asterisk.service - LSB: Asterisk PBX...
juin 20 08:31:46 pbx asterisk[2361]: Starting Asterisk PBX: asterisk.
juin 20 08:31:46 pbx systemd[1]: Started asterisk.service - LSB: Asterisk PBX.
root@pbx:/usr/src/asterisk-20.3.0#
```

On va faire une sauvegarde des fichiers que l'on va modifier :

mv /etc/asterisk/pjsip.conf /etc/asterisk/pjsip.conf.origin
mv /etc/asterisk/voicemail.conf /etc/asterisk/voicemail.conf.origin
cp /etc/asterisk/asterisk.conf /etc/asterisk/asterisk.conf.origin

Pour le test on va créer 3 Utilisateurs :
alice, numéro : 6001, mot de passe : bonjour
bob, numéro : 6002, mot de passe : bonjour
martin, numéro : 6003, mot de passe : bonjour

Asterisk – From Scratch

Les fichiers de config sont a récupérer sur : https://github.com/thierry-rami/Asterisk_config

Une fois les fichiers de config installés faire :
asterisk -rx "core restart now"

et on vérifie que Astesrisk Tourne :

```
root@pbx:~# ss -nlut | grep -E '5060|State'
Netid  State   Recv-Q  Send-Q   Local Address:Port    Peer Address:Port
udp    UNCONN  0       0        0.0.0.0:5060          0.0.0.0:*
```

On lance la commande asterisk -rvvvvvvvvvv
on se trouve maintenant dans l'invite de commande de asterisk :

pjsip list aors
pjsip list auths

Normalement on devrait voir les 3 utilisateurs : Alice , Bob , Martin

Asterisk – From Scratch

Maintenant que le serveur Asterisk tourne , on va installer le client sur Windows , Linux , MacOs , Smartphone : Linphone

Dans ce cas je me connecte en tant que « Alice »
et l'adresse IP du serveur asterisk est:10.10.18.150
Une fois connecté dans l'interface CLI d'Asterisk on a :

```
    -- Added contact 'sip:alice@10.10.18.151:44678;transport=udp' to AOR 'alice' with expiration of 3600 seconds
  == Endpoint alice is now Reachable
    -- Removed contact 'sip:alice@10.10.18.151:44678;transport=udp' from AOR 'alice' due to request
  == Contact alice/sip:alice@10.10.18.151:44678;transport=udp has been deleted
  == Endpoint alice is now Unreachable
pbx*CLI>
```