

**PILOTO DE SISTEMA DE COMUNICACIÓN CON VoWLAN CONTROLADO
POR COMANDOS DE VOZ PARA UN ENTORNO HOSPITALARIO.**



ANEXO C

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ANEXO C: PRUEBAS DE COMPATIBILIDAD Y LA CALIDAD DE LA RECEPCIÓN DEL AUDIO DE CADA UNO DE LOS CÓDECS DEL SERVIDOR CUANDO SE VARÍAN LOS DEL CLIENTE

Debido a que en el servidor y los usuarios manejan *softphones* diferentes, dado que están basados en sistemas operativos diferentes (Linux y Windows, respectivamente) se realizan pruebas de compatibilidad entre los *codecs* de cada uno de ellos, de esta manera trabajar con las parejas que permitan una entrega y recepción adecuada de los paquetes de voz.

CODECS DE SOFTPHONE X-LITE (CLIENTES):

- Broadvoice-32
- Broadvoice-32FEC
- DVI4
- DVI4 Wideband
- ITU G711 ulaw
- GSM
- iLBC
- L16PCM Wideband
- ITU G711 alaw

CODECS DE SOFTPHONE SJPHONE (SERVIDOR):

- GSM 6.10
- iLBC-30ms
- iLBC-20ms
- Speex 15.2K
- Speex 15.2K 40 ms
- Speex 8.0K 40 ms
- ITU G711 alaw
- ITU G711 ulaw

En la figura C1 se puede observar al codec Broadvoice-32 del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del sphone:

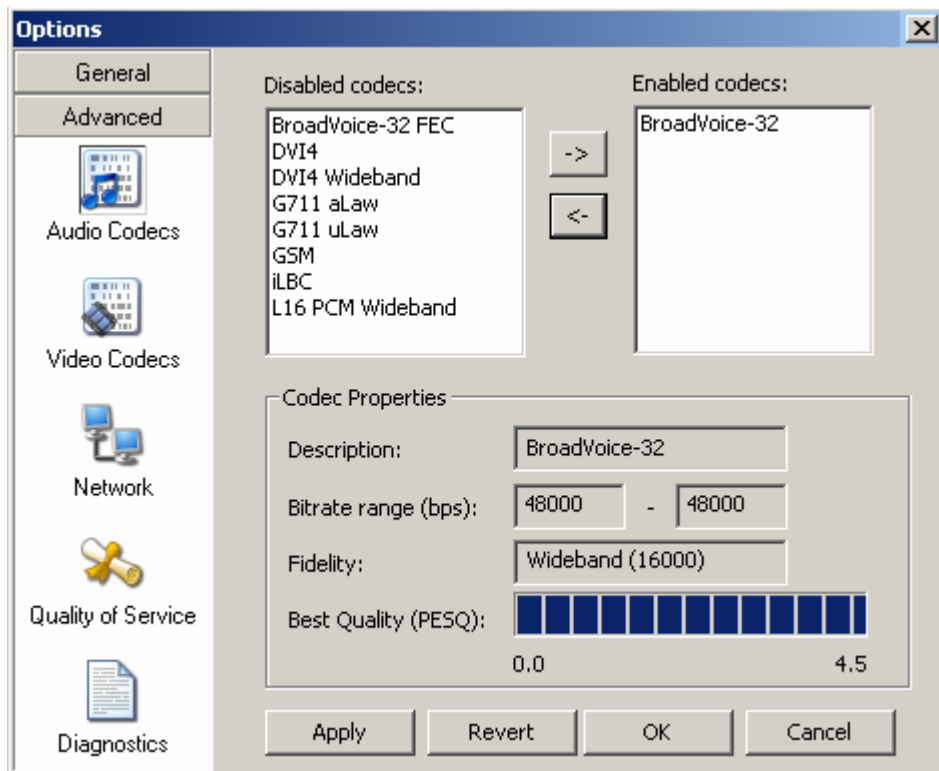
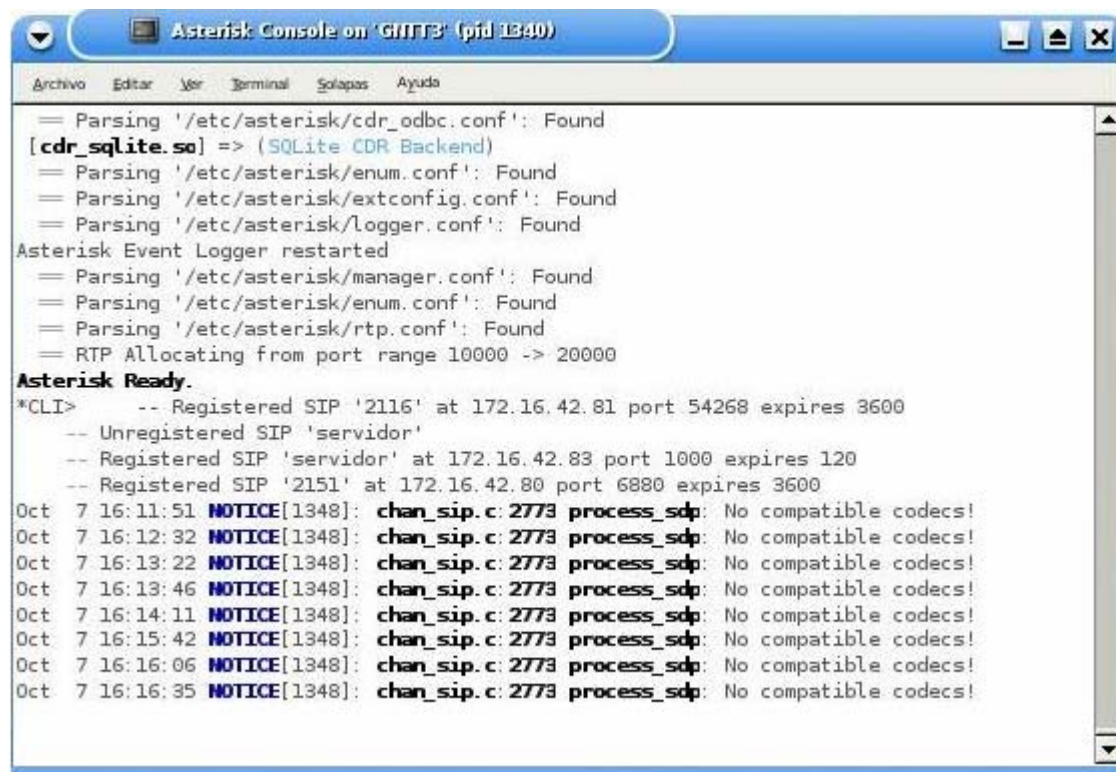


Figura C1. Softphone X-lite con el codec Broadvoice-32 habilitado.

- GSM 6.10 no compatibles
- iLBC-30ms no compatibles
- iLBC-20ms no compatibles
- Speex 15.2K no compatibles
- Speex 15.2K 40 ms no compatibles
- Speex 8.0K 40 ms no compatibles
- ITU G711 alaw no compatibles
- ITU G711 ulaw no compatibles

En la figura C2 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'GIMP3' (pid 1340)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda
= Parsing '/etc/asterisk/cdr_odbc.conf': Found
[ cdr_sqlite.so ] => (SQLite CDR Backend)
= Parsing '/etc/asterisk/enum.conf': Found
= Parsing '/etc/asterisk/extconfig.conf': Found
= Parsing '/etc/asterisk/logger.conf': Found
Asterisk Event Logger restarted
= Parsing '/etc/asterisk/manager.conf': Found
= Parsing '/etc/asterisk/enum.conf': Found
= Parsing '/etc/asterisk/rtp.conf': Found
= RTP Allocating from port range 10000 -> 20000
Asterisk Ready.
*CLI> -- Registered SIP '2116' at 172.16.42.81 port 54268 expires 3600
-- Unregistered SIP 'servidor'
-- Registered SIP 'servidor' at 172.16.42.83 port 1000 expires 120
-- Registered SIP '2151' at 172.16.42.80 port 6880 expires 3600
Oct 7 16:11:51 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:12:32 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:13:22 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:13:46 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:14:11 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:15:42 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:16:06 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
Oct 7 16:16:35 NOTICE[1348]: chan_sip.c:2773 process_sdp: No compatible codecs!
```

Figura C2. Resultado de la prueba con el codec Broadvoice-32

En la figura C3 se puede observar al codec Broadvoice-32 FEC del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del siphone:

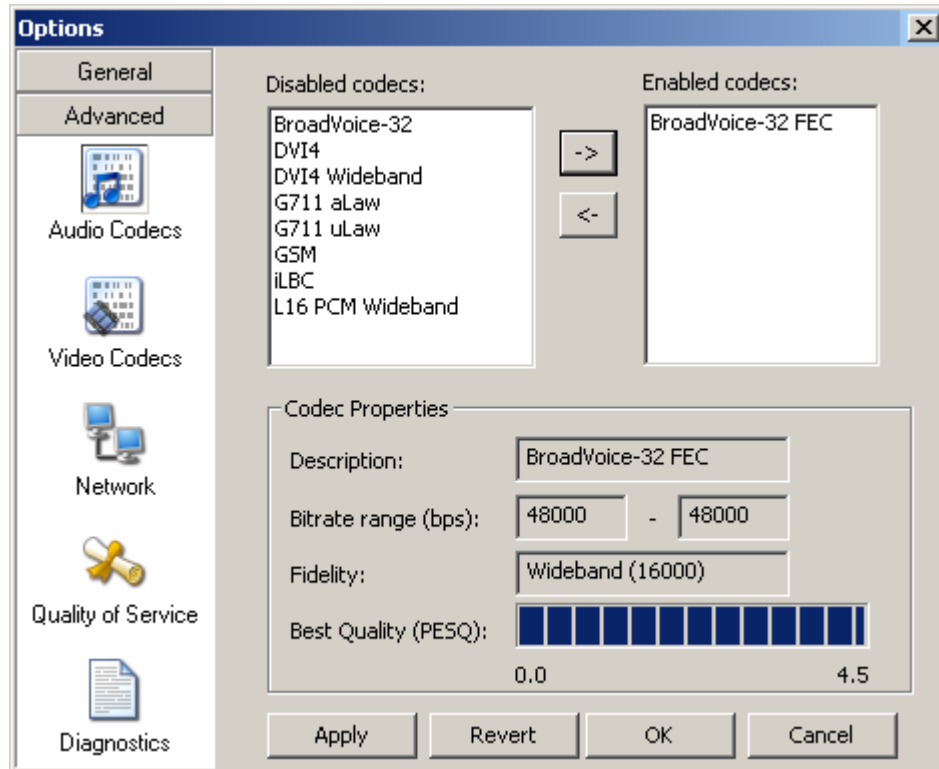
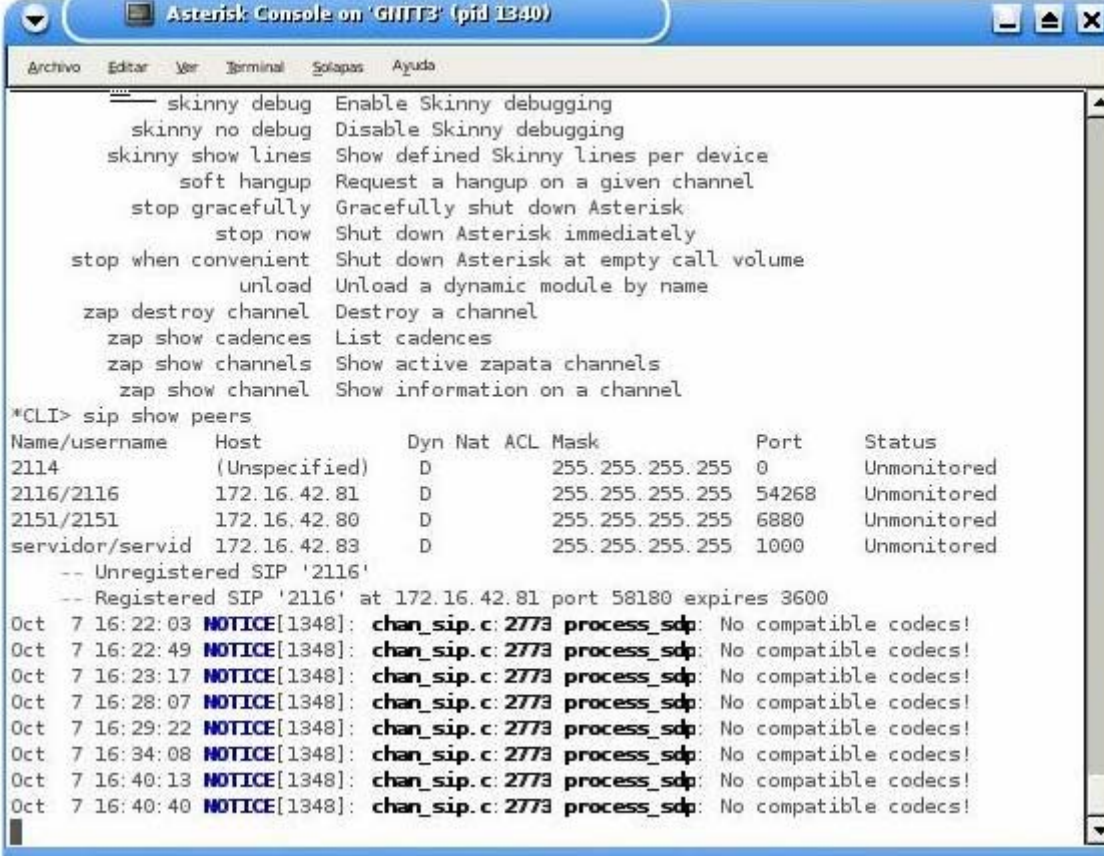


Figura C3. Softphone X-lite con solo el codec Broadvoice-32 FEC habilitado

- GSM 6.10 no compatibles
- iLBC-30ms no compatibles
- iLBC-20ms no compatibles
- Speex 15.2K no compatibles
- Speex 15.2K 40 ms no compatibles
- Speex 8.0K 40 ms no compatibles
- ITU G711 alaw no compatibles
- ITU G711 ulaw no compatibles

En la figura C4 se ve el resultado de la prueba en la consola de Asterisk.



```
----- skinny debug Enable Skinny debugging
skinny no debug Disable Skinny debugging
skinny show lines Show defined Skinny lines per device
soft hangup Request a hangup on a given channel
stop gracefully Gracefully shut down Asterisk
stop now Shut down Asterisk immediately
stop when convenient Shut down Asterisk at empty call volume
unload Unload a dynamic module by name
zap destroy channel Destroy a channel
zap show cadences List cadences
zap show channels Show active zapata channels
zap show channel Show information on a channel

*CLI> sip show peers
Name/username Host Dyn Nat ACL Mask Port Status
2114 (Unspecified) D 255.255.255.255 0 Unmonitored
2116/2116 172.16.42.81 D 255.255.255.255 54268 Unmonitored
2151/2151 172.16.42.80 D 255.255.255.255 6880 Unmonitored
servidor/servid 172.16.42.83 D 255.255.255.255 1000 Unmonitored

-- Unregistered SIP '2116'
-- Registered SIP '2116' at 172.16.42.81 port 58180 expires 3600
Oct 7 16:22:03 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:22:49 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:23:17 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:28:07 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:29:22 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:34:08 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:40:13 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct 7 16:40:40 NOTICE[1348]: chan_sip.c: 2773 process_sdp: No compatible codecs!
```

Figura C4. Resultado de la prueba con el codec Broadvoice-32 FEC en la consola de Asterisk.

En la figura C5 se puede observar al codec DVI4 del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del softphone:

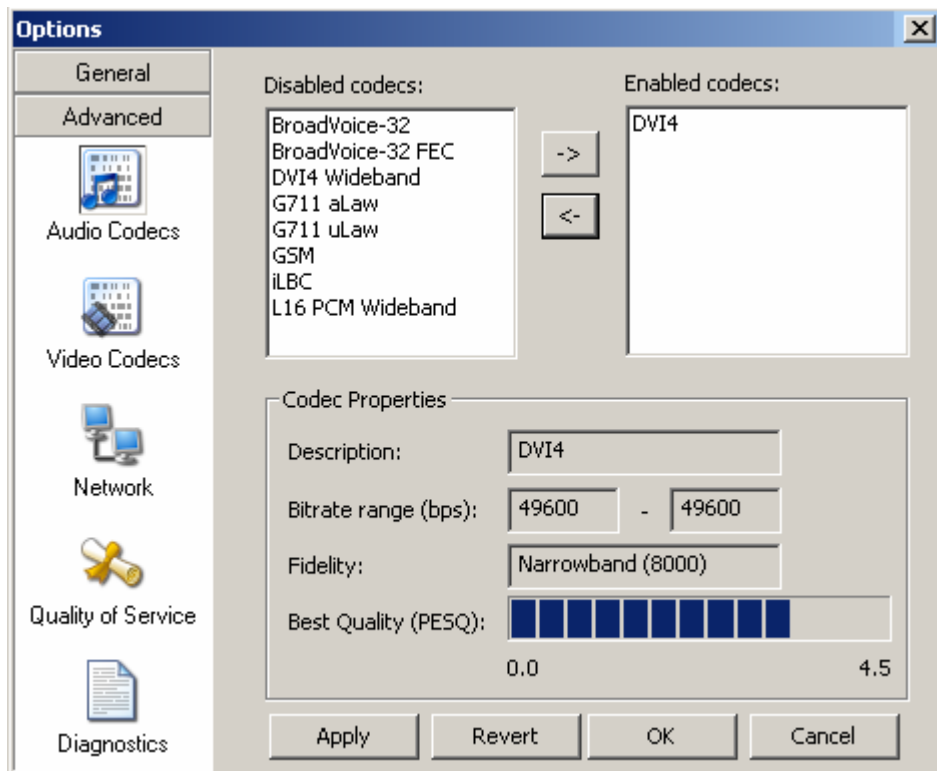


Figura C5. Softphone X-lite con solo el codec DVI4 habilitado.

- -GSM 6.10: la llamada se establece pero se escucha mucho ruido.
- -iLBC-30ms: la llamada se genera pero no se establece, en la figura C6 se observa lo que aparece en la interfaz del cliente.
- -iLBC-20ms: sucede lo mismo que con el iLBC-30ms
- -Speex 15.2K: la llamada se establece pero hay mucho ruido que evita que se escuche claramente.
- -Speex 15.2K 40 ms:sucede lo mismo que con el Speex 15.2K.
- -Speex 8.0K 40 ms: La llamada se establece pero el ruido es mayor que con los codecs anteriores, se escucha muy mal.
- -ITU G711 alaw: La llamada se establece pero se escucha muy lejos y con mucho ruido.
- -ITU G711 ulaw: La llamada se establece pero se escucha muy lejos y con mucho ruido.



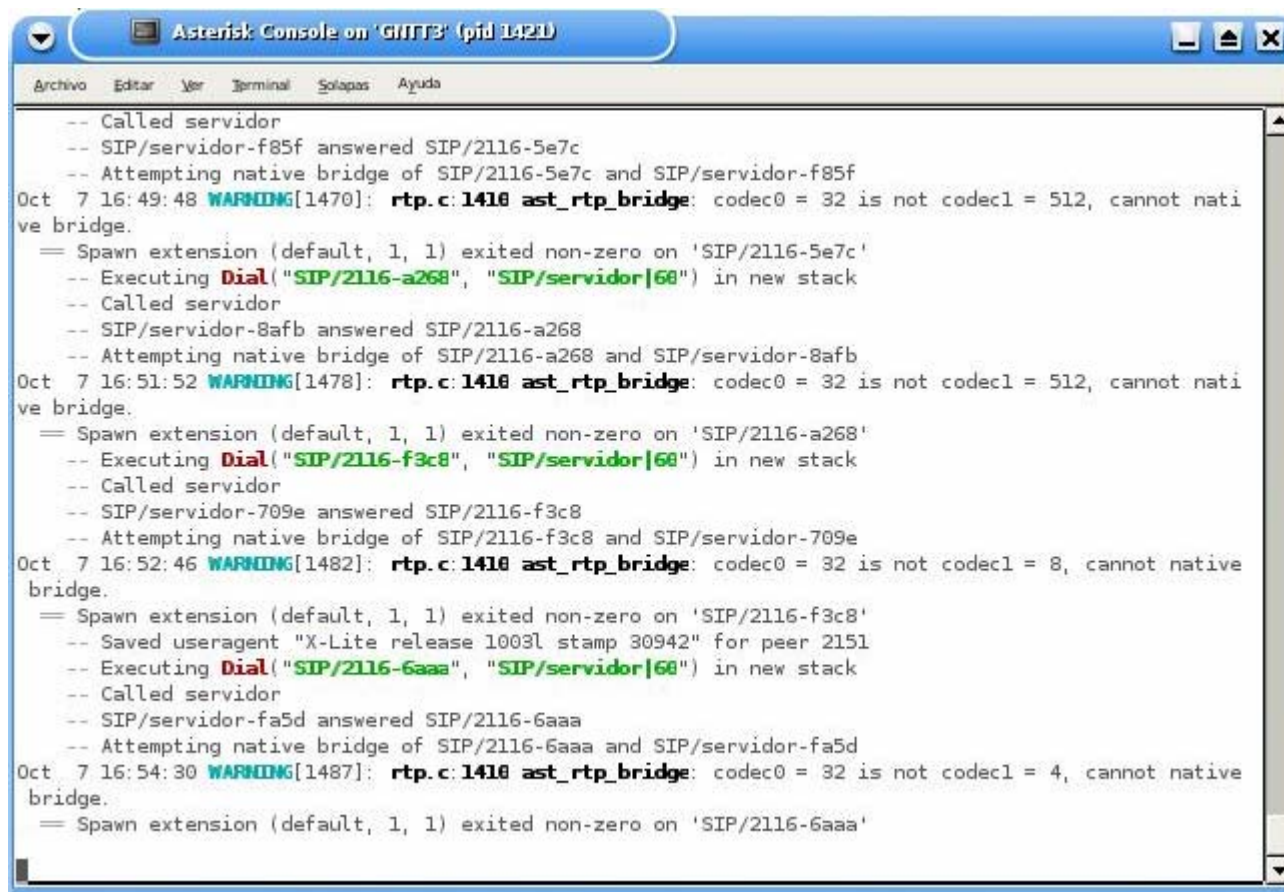
Figura C6. Softphone del cliente cuando no se establece una llamada

En las figuras C7 y C8 se ve el resultado de la prueba en la consola de Asterisk.

```
Asterisk Console on 'GIIIT3' (pid 1421)
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

Asterisk Ready.
*CLI> -- Executing Dial("SIP/2116-c9d2", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-cdf9 answered SIP/2116-c9d2
-- Attempting native bridge of SIP/2116-c9d2 and SIP/servidor-cdf9
Oct 7 16:44:25 WARNING[1442]: rtp.c: 1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 2, cannot native
bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-c9d2'
-- Executing Dial("SIP/2116-3ac3", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 16:45:16 WARNING[1446]: pbx.c: 1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-7cd1", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 16:45:31 WARNING[1449]: pbx.c: 1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-772c", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-35a4 answered SIP/2116-772c
-- Attempting native bridge of SIP/2116-772c and SIP/servidor-35a4
Oct 7 16:47:26 WARNING[1442]: rtp.c: 1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 2, cannot native
```

Figura C7. Resultado de la prueba con el codec DVI4 en la consola de Asterisk



```
Asterisk Console on 'GINT3' (pid 1421)
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

-- Called servidor
-- SIP/servidor-f85f answered SIP/2116-5e7c
-- Attempting native bridge of SIP/2116-5e7c and SIP/servidor-f85f
Oct 7 16:49:48 WARNING[1470]: rtp.c:1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-5e7c'
-- Executing Dial("SIP/2116-a268", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-8afb answered SIP/2116-a268
-- Attempting native bridge of SIP/2116-a268 and SIP/servidor-8afb
Oct 7 16:51:52 WARNING[1478]: rtp.c:1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-a268'
-- Executing Dial("SIP/2116-f3c8", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-709e answered SIP/2116-f3c8
-- Attempting native bridge of SIP/2116-f3c8 and SIP/servidor-709e
Oct 7 16:52:46 WARNING[1482]: rtp.c:1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 8, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-f3c8'
-- Saved useragent "X-Lite release 1003l stamp 30942" for peer 2151
-- Executing Dial("SIP/2116-6aaa", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-fa5d answered SIP/2116-6aaa
-- Attempting native bridge of SIP/2116-6aaa and SIP/servidor-fa5d
Oct 7 16:54:30 WARNING[1487]: rtp.c:1416 ast_rtp_bridge: codec0 = 32 is not codec1 = 4, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-6aaa'
```

Figura C8. Resultado de la prueba con el codec DVI4 en la consola de Asterisk

En la figura C9 se puede observar al codec DVI4 Wideband del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del siphone:

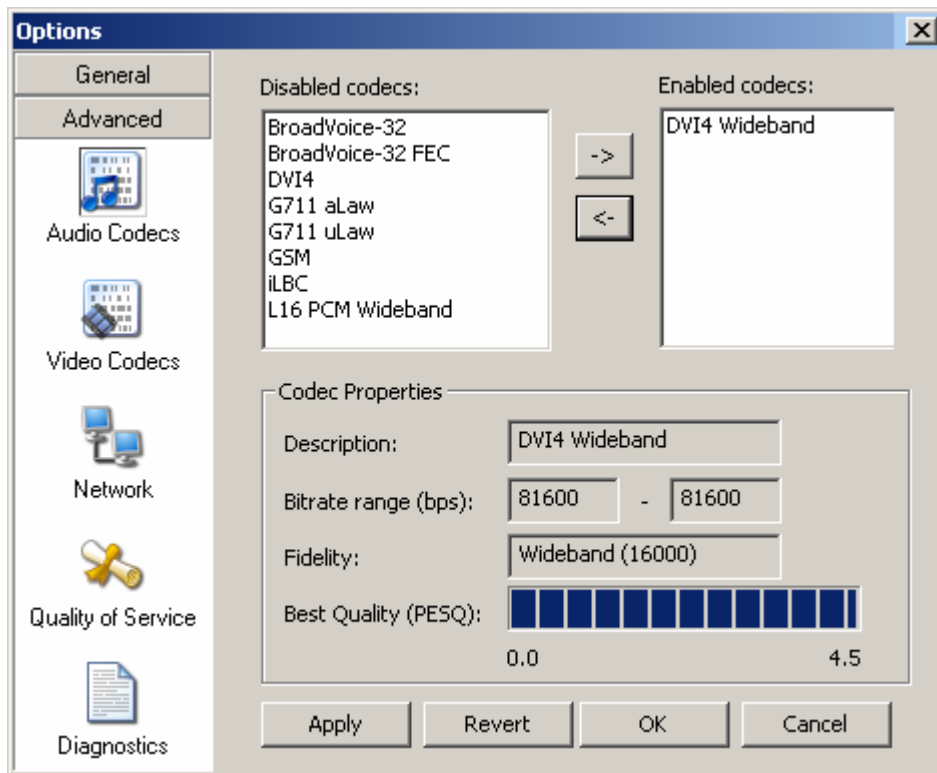
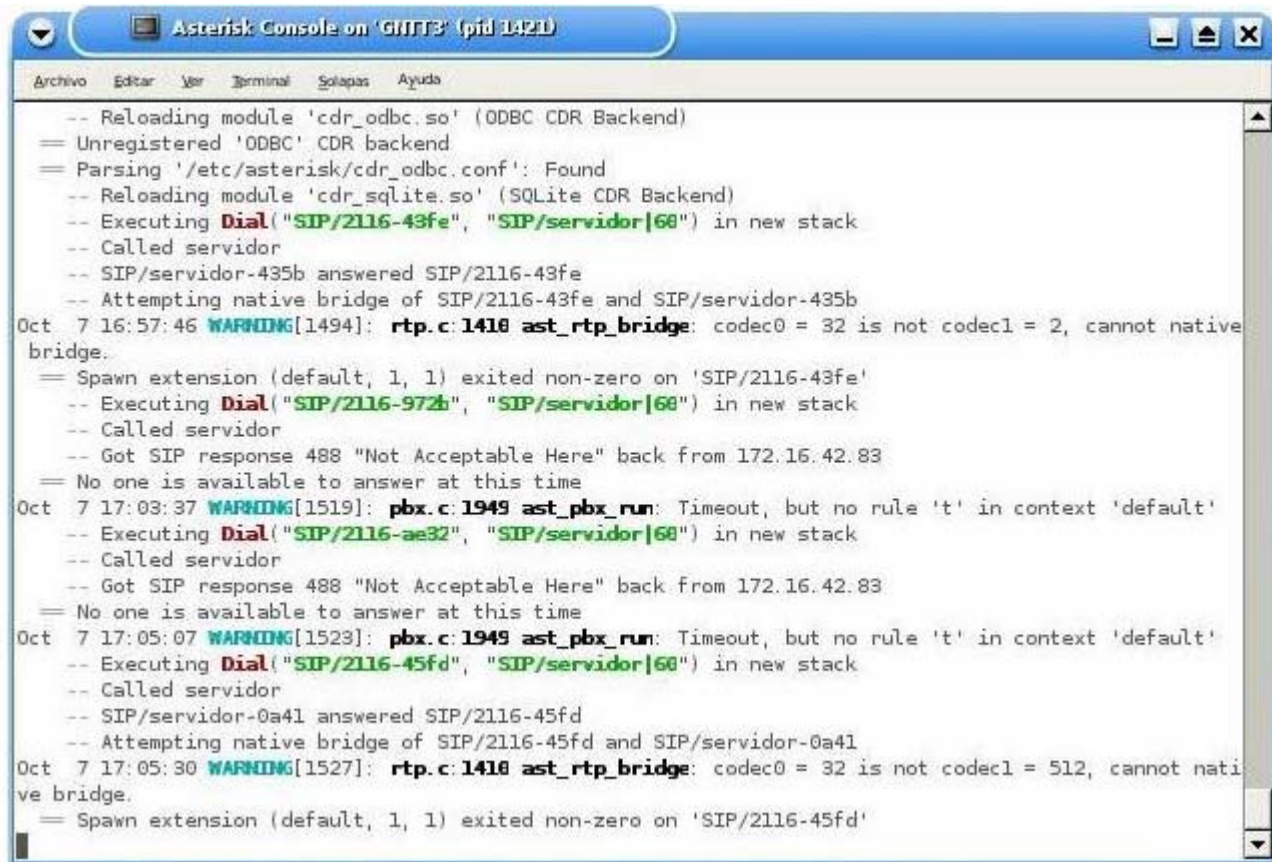


Figura C9. Softphone X-lite con solo el codec DVI4 Wideband habilitado

- GSM 6.10: La llamada se establece, la voz se escucha poco debido al ruido.
- iLBC-30ms: La llamada se genera pero no se establece.
- iLBC-20ms La llamada se genera pero no se establece.
- -Speex 15.2K: La llamada se establece, pero el ruido hace que no se entienda claramente la voz.
- Speex 15.2K 40 ms: Sucede lo mismo que con el Speex 15.2K.
- Speex 8.0K 40 ms: Sucede lo mismo que con el Speex 15.2K
- ITU G711 alaw: La llamada se establece, la voz es un poco más entendible que con los otros codecs, sin embargo sigue existiendo ruido.
- ITU G711 ulaw: lo mismo que con el ITU G711alaw.

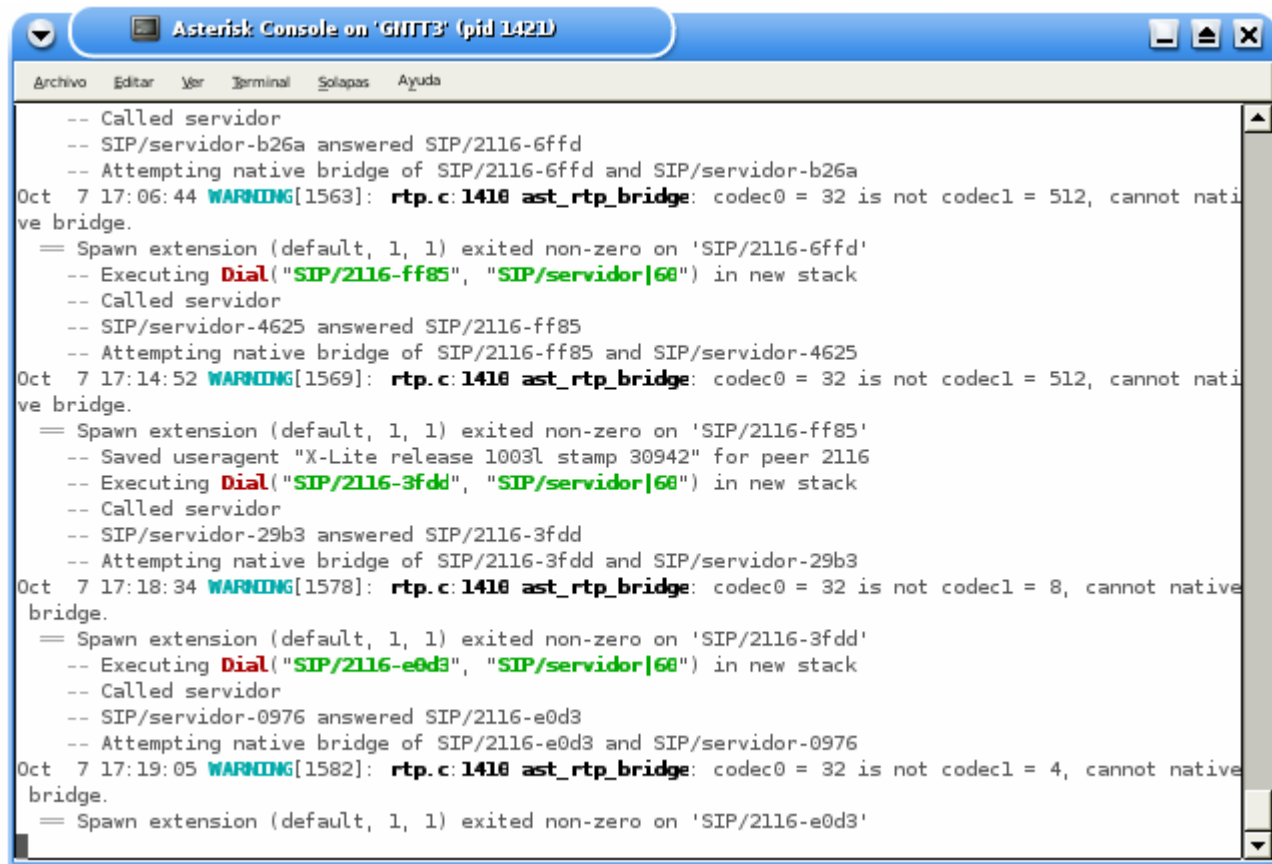
En las figuras C10 y C11 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'GHTT3' (pid 1421)
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
== Unregistered 'ODBC' CDR backend
== Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-43fe", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-435b answered SIP/2116-43fe
-- Attempting native bridge of SIP/2116-43fe and SIP/servidor-435b
Oct 7 16:57:46 WARNING[1494]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 2, cannot native
bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-43fe'
-- Executing Dial("SIP/2116-972b", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 17:03:37 WARNING[1519]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-ae32", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 17:05:07 WARNING[1523]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-45fd", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-0a41 answered SIP/2116-45fd
-- Attempting native bridge of SIP/2116-45fd and SIP/servidor-0a41
Oct 7 17:05:30 WARNING[1527]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 512, cannot nati
ve bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-45fd'
```

Figura C10. Resultado de la prueba con el codec DVI4 Wideband en la consola de Asterisk



```
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

-- Called servidor
-- SIP/servidor-b26a answered SIP/2116-6ffd
-- Attempting native bridge of SIP/2116-6ffd and SIP/servidor-b26a
Oct  7 17:06:44 WARNING[1563]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 512, cannot native
ve bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-6ffd'
-- Executing Dial("SIP/2116-ff85", "SIP/servidor|68") in new stack
-- Called servidor
-- SIP/servidor-4625 answered SIP/2116-ff85
-- Attempting native bridge of SIP/2116-ff85 and SIP/servidor-4625
Oct  7 17:14:52 WARNING[1569]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 512, cannot native
ve bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-ff85'
-- Saved useragent "X-Lite release 1003l stamp 30942" for peer 2116
-- Executing Dial("SIP/2116-3fdd", "SIP/servidor|68") in new stack
-- Called servidor
-- SIP/servidor-29b3 answered SIP/2116-3fdd
-- Attempting native bridge of SIP/2116-3fdd and SIP/servidor-29b3
Oct  7 17:18:34 WARNING[1578]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 8, cannot native
bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-3fdd'
-- Executing Dial("SIP/2116-e0d3", "SIP/servidor|68") in new stack
-- Called servidor
-- SIP/servidor-0976 answered SIP/2116-e0d3
-- Attempting native bridge of SIP/2116-e0d3 and SIP/servidor-0976
Oct  7 17:19:05 WARNING[1582]: rtp.c:1418 ast_rtp_bridge: codec0 = 32 is not codec1 = 4, cannot native
bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-e0d3'
```

Figura C11. Resultado de la prueba con el codec DVI4 Wideband

En la figura C12 se puede observar al codec ITU G711 ulaw del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del softphone:

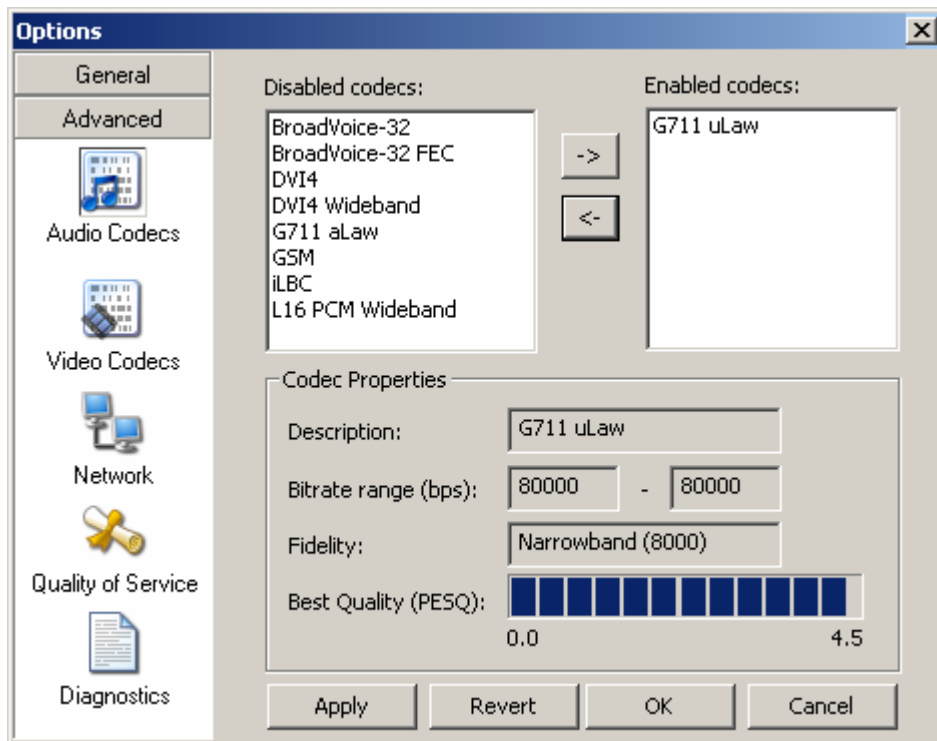


Figura C12. Softphone X-lite con solo el codec ITU G711 ulaw habilitado

- GSM 6.10: La llamada se establece, se presenta un poco de retardo y el ruido ya no es perceptible.
- iLBC-30ms: La llamada se genera pero no se establece.
- iLBC-20ms La llamada se genera pero no se establece.
- Speex 15.2K: La llamada se establece, pero la voz se escucha cortada.
- Speex 15.2K 40 ms: La llamada se establece, la voz se escucha mas claramente que con el Speex 15.2K.
- Speex 8.0K 40 ms: La llamada se establece, la voz se escucha claro pero presenta retardo significativo. Cuando se habla de retardo significativo o considerable es de mas de 500ms.
- ITU G711 alaw: La llamada se establece, la voz es clara, entendible y se presenta poco retardo.
- ITU G711 ulaw: La llamada se establece, la voz es clara, entendible pero se presenta un retardo significativo.

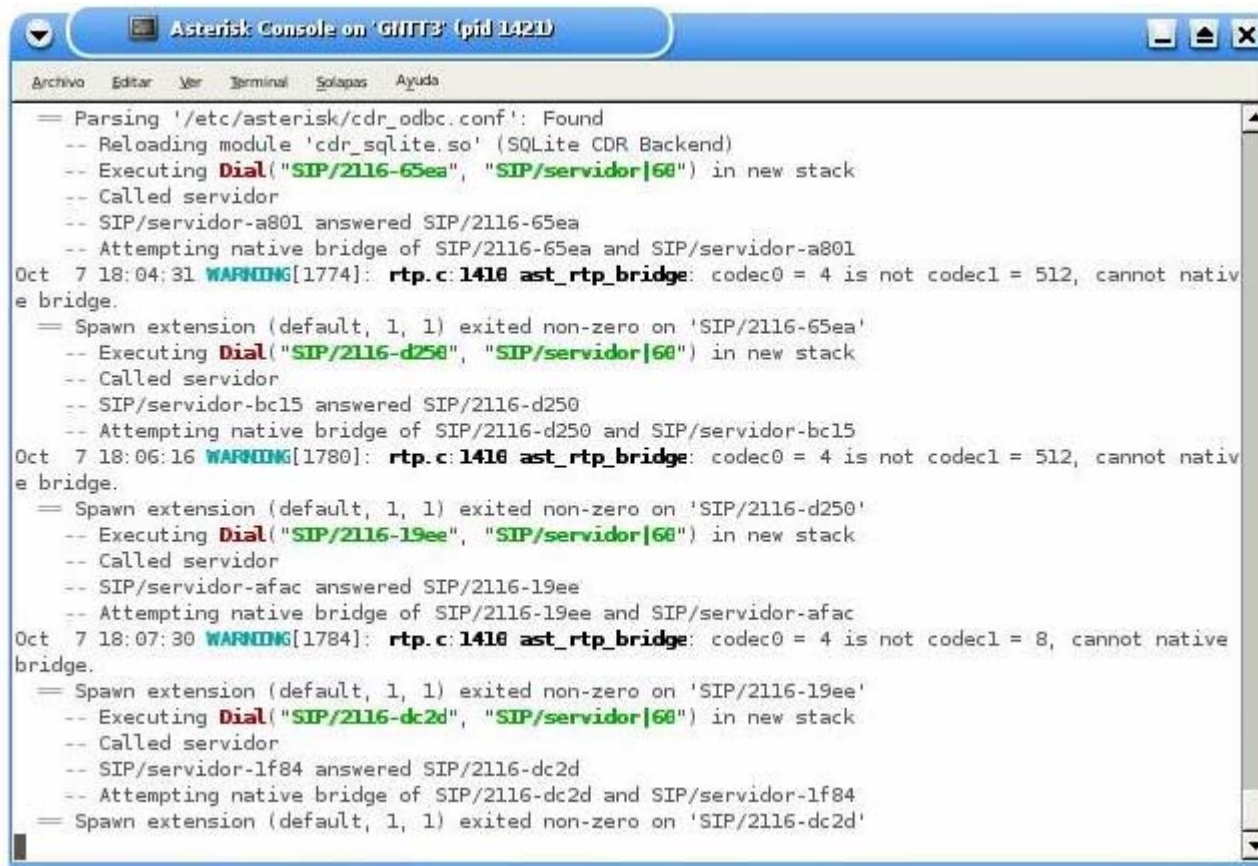
En las figuras C13 y C14 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'GITE' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda

-- Executing Dial("SIP/2116-b657", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
= No one is available to answer at this time
Oct 7 17:51:10 WARNING[1741]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-c744", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
= No one is available to answer at this time
Oct 7 17:51:51 WARNING[1745]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-fc9b", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-8369 answered SIP/2116-fc9b
-- Attempting native bridge of SIP/2116-fc9b and SIP/servidor-8369
Oct 7 17:54:06 WARNING[1749]: rtp.c:1418 ast_rtp_bridge: codec0 = 4 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-fc9b'
-- Executing Dial("SIP/2116-f66d", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-1a5b answered SIP/2116-f66d
-- Attempting native bridge of SIP/2116-f66d and SIP/servidor-1a5b
Oct 7 18:03:20 WARNING[1758]: rtp.c:1418 ast_rtp_bridge: codec0 = 4 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-f66d'
```

Figura C13. Resultado de la prueba con el codec ITU G711 ulaw



```
Asterisk Console on 'GITE' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda

= Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-65ea", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-a801 answered SIP/2116-65ea
-- Attempting native bridge of SIP/2116-65ea and SIP/servidor-a801
Oct 7 18:04:31 WARNING[1774]: rtp.c:1418 ast_rtp_bridge: codec0 = 4 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-65ea'
-- Executing Dial("SIP/2116-d250", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-bc15 answered SIP/2116-d250
-- Attempting native bridge of SIP/2116-d250 and SIP/servidor-bc15
Oct 7 18:06:16 WARNING[1780]: rtp.c:1418 ast_rtp_bridge: codec0 = 4 is not codec1 = 512, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-d250'
-- Executing Dial("SIP/2116-19ee", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-afac answered SIP/2116-19ee
-- Attempting native bridge of SIP/2116-19ee and SIP/servidor-afac
Oct 7 18:07:30 WARNING[1784]: rtp.c:1418 ast_rtp_bridge: codec0 = 4 is not codec1 = 8, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-19ee'
-- Executing Dial("SIP/2116-dc2d", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-1f84 answered SIP/2116-dc2d
-- Attempting native bridge of SIP/2116-dc2d and SIP/servidor-1f84
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-dc2d'
```

Figura C14. Resultado de la prueba con el codec ITU G711 ulaw

En la figura C15 se puede observar al codec GSM del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del siphone:

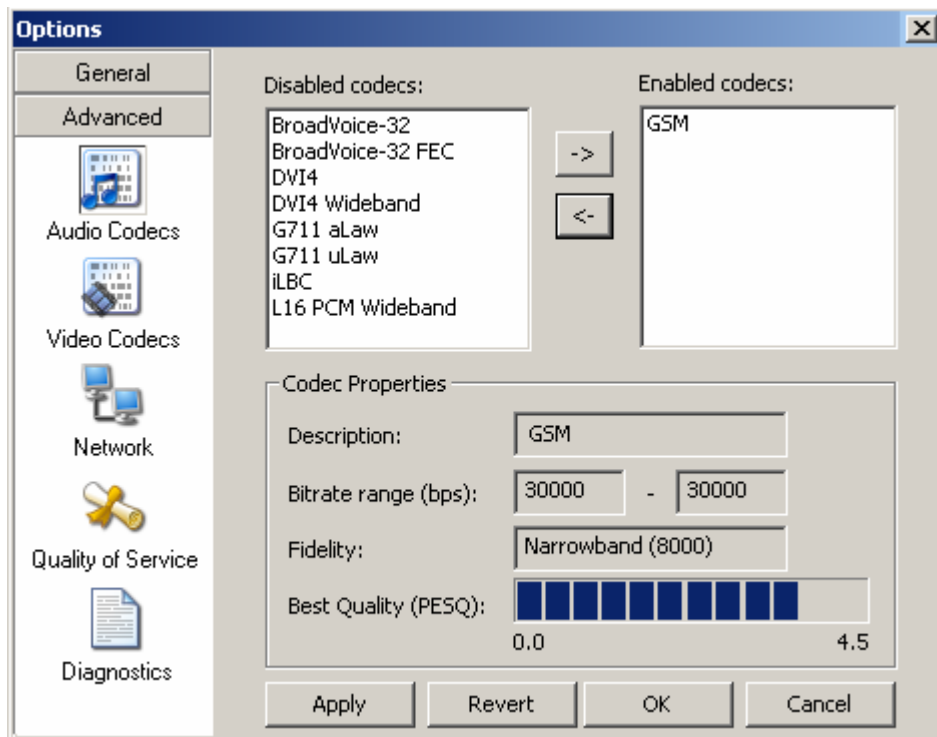
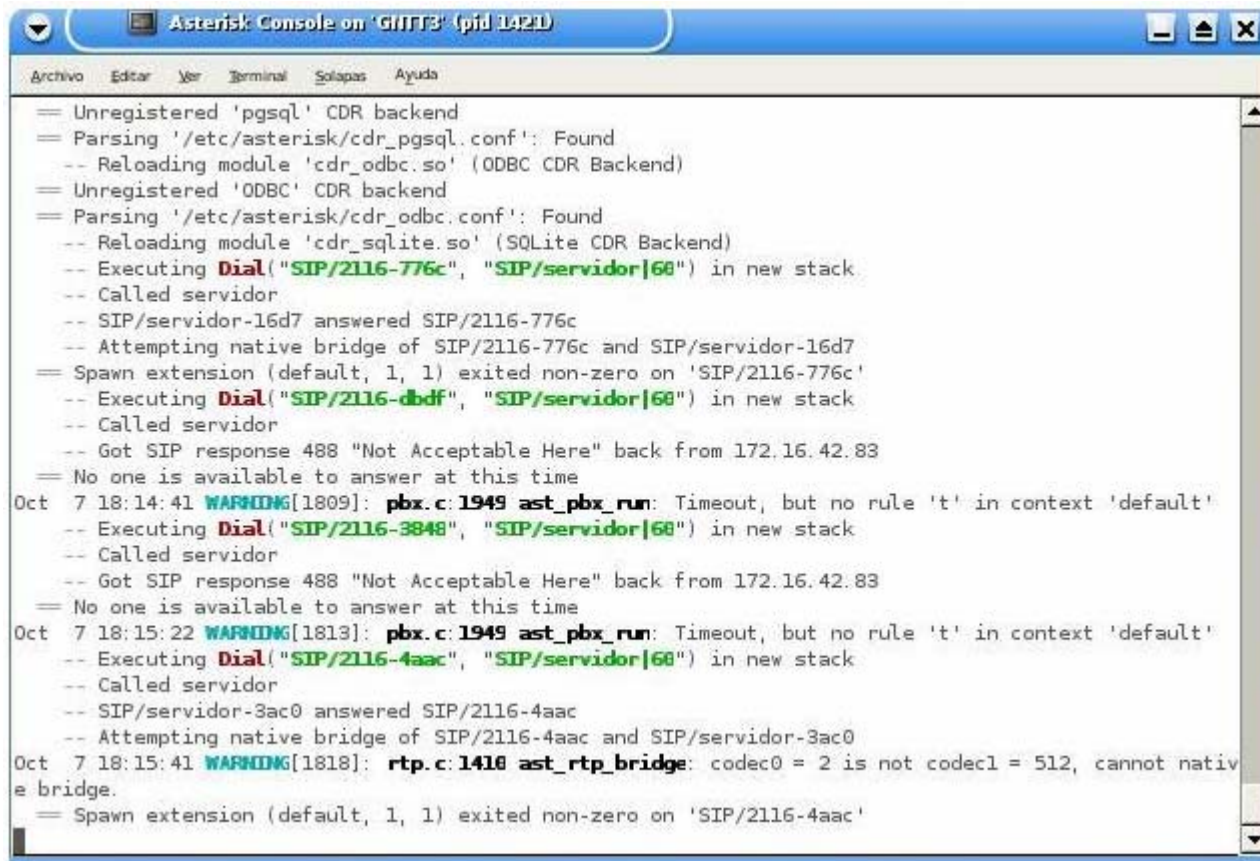


Figura C15. Softphone X-lite con solo el codec GSM Wideband habilitado

- GSM 6.10: La llamada se establece, la voz es entendible y el retardo no es muy significativo.
- iLBC-30ms: La llamada se genera pero no se establece.
- iLBC-20ms La llamada se genera pero no se establece.
- Speex 15.2K: La llamada se establece, se presenta retardo considerable y la voz se escucha lejos (poco volumen).
- Speex 15.2K 40 ms: La llamada se establece, la voz no se entiende debido a que se presenta bastante ruido.
- Speex 8.0K 40 ms: La llamada se establece, la voz se escucha un poco mejor que con el Speex 15.2k de 40 ms pero presenta retardo considerable.
- ITU G711 alaw: La llamada se establece, la voz es clara, entendible y se presenta poco retardo.
- ITU G711 ulaw: La llamada se establece, la voz es clara, entendible pero se presenta un retardo significativo.

En las figuras C16 y C17 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'GIMP3' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda

== Unregistered 'pgsql' CDR backend
== Parsing '/etc/asterisk/cdr_pgsql.conf': Found
-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
== Unregistered 'ODBC' CDR backend
== Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-776c", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-16d7 answered SIP/2116-776c
-- Attempting native bridge of SIP/2116-776c and SIP/servidor-16d7
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-776c'
-- Executing Dial("SIP/2116-dbd", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 18:14:41 WARNING[1809]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-3848", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 18:15:22 WARNING[1813]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-4aac", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-3ac0 answered SIP/2116-4aac
-- Attempting native bridge of SIP/2116-4aac and SIP/servidor-3ac0
Oct 7 18:15:41 WARNING[1818]: rtp.c:1418 ast_rtp_bridge: codec0 = 2 is not codec1 = 512, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-4aac'
```

Figura C16. Resultado de la prueba con el codec GSM en la consola de Asterisk



```
Asterisk Console on 'GITE' (pid 1421)
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

-- Executing Dial("SIP/2116-5630", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-0477 answered SIP/2116-5630
-- Attempting native bridge of SIP/2116-5630 and SIP/servidor-0477
Oct 7 18:16:56 WARNING[1845]: rtp.c:1418 ast_rtp_bridge: codec0 = 2 is not codec1 = 512, cannot native
e bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-5630'
-- Executing Dial("SIP/2116-7bd4", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-bba7 answered SIP/2116-7bd4
-- Attempting native bridge of SIP/2116-7bd4 and SIP/servidor-bba7
Oct 7 18:18:18 WARNING[1851]: rtp.c:1418 ast_rtp_bridge: codec0 = 2 is not codec1 = 512, cannot native
e bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-7bd4'
-- Executing Dial("SIP/2116-4acd", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-99e1 answered SIP/2116-4acd
-- Attempting native bridge of SIP/2116-4acd and SIP/servidor-99e1
Oct 7 18:19:28 WARNING[1855]: rtp.c:1418 ast_rtp_bridge: codec0 = 2 is not codec1 = 8, cannot native
bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-4acd'
-- Executing Dial("SIP/2116-4257", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-d281 answered SIP/2116-4257
-- Attempting native bridge of SIP/2116-4257 and SIP/servidor-d281
Oct 7 18:20:28 WARNING[1859]: rtp.c:1418 ast_rtp_bridge: codec0 = 2 is not codec1 = 4, cannot native
bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-4257'
```

Figura C17. Resultado de la prueba con el codec GSM en la consola de Asterisk

En la figura C18 se puede observar al codec iLBC del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del softphone:

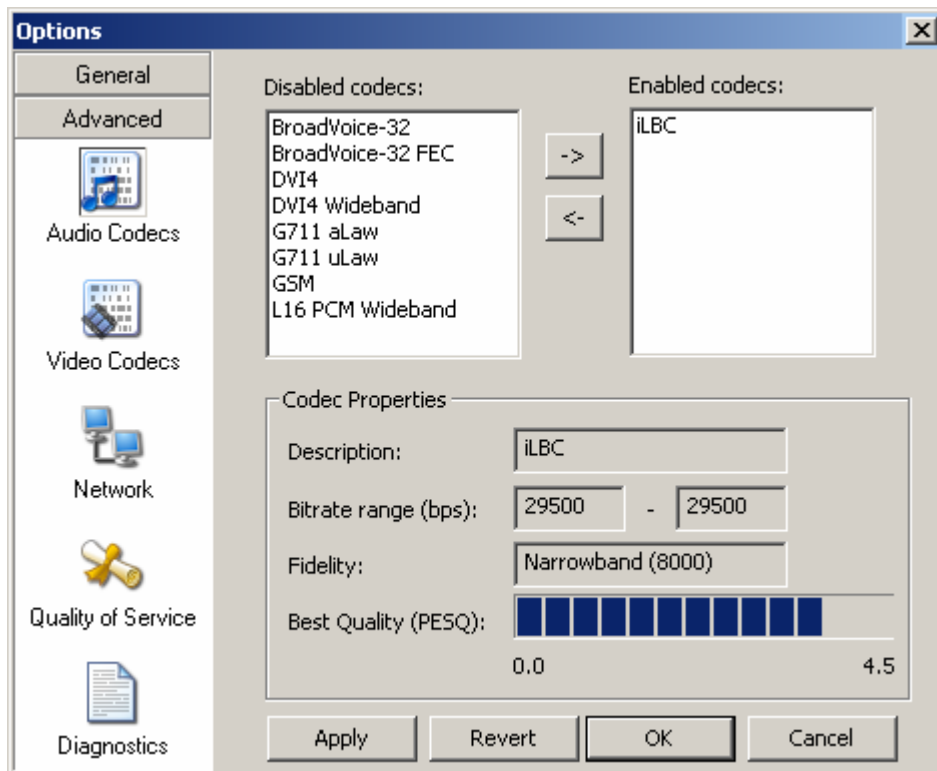
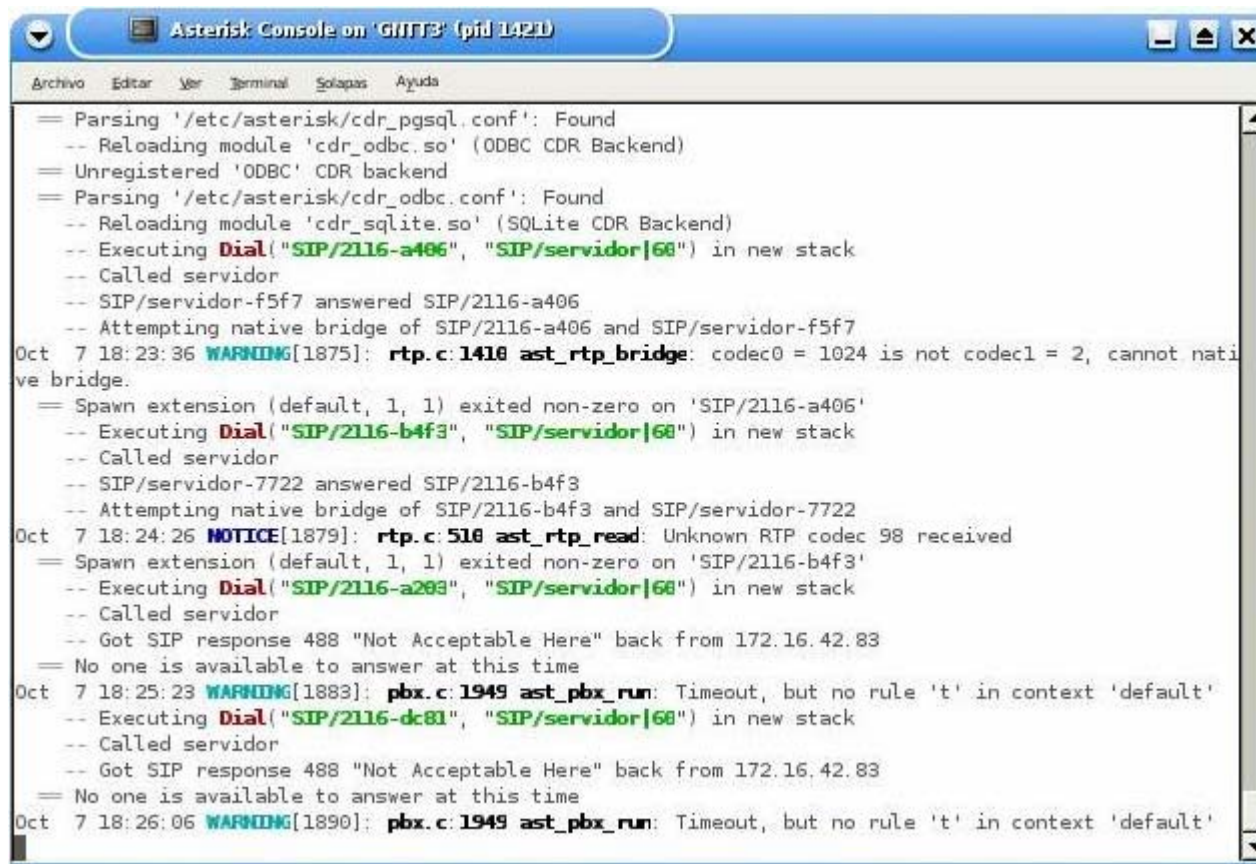


Figura C18. Softphone X-lite con solo el codec iLBC Wideband habilitado

- GSM 6.10: La llamada se establece, la voz es entendible y el retardo no es muy significativo.
- iLBC-30ms: La llamada se establece pero no se escucha la voz en el receptor.
- iLBC-20ms: La llamada se genera pero no se establece.
- Speex 15.2K: La llamada se genera pero no se establece.
- Speex 15.2K 40 ms: La llamada se genera pero no se establece.
- Speex 8.0K 40 ms: La llamada se genera pero no se establece.
- ITU G711 alaw: La llamada se establece, la voz es clara, entendible pero el retardo es significativo.
- ITU G711 ulaw: La llamada se establece, la voz es clara, entendible y se presenta un retardo menor que con el codec ITU G711 alaw.

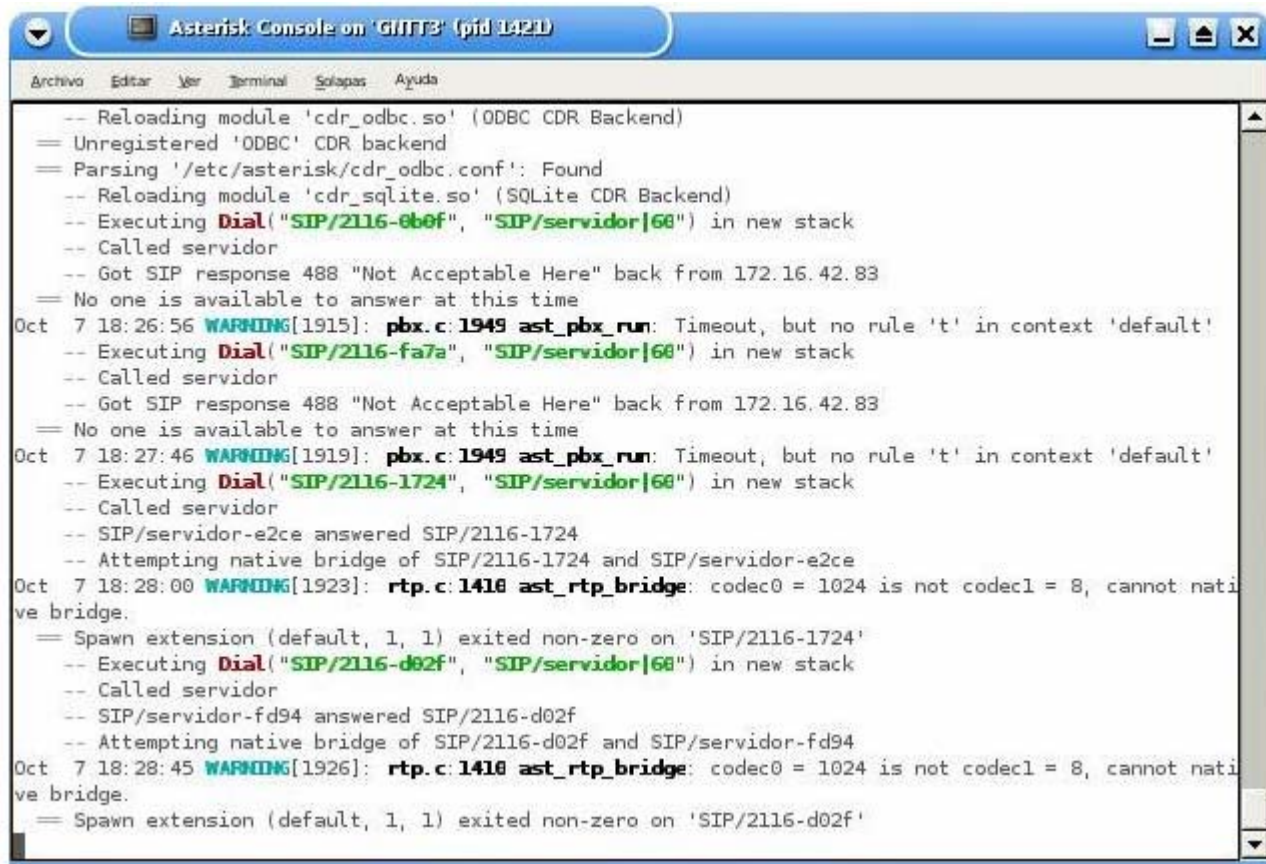
En las figuras C19 y C20 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'GIMP3' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda

= Parsing '/etc/asterisk/cdr_pgsqll.conf': Found
-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
= Unregistered 'ODBC' CDR backend
= Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-a406", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-f5f7 answered SIP/2116-a406
-- Attempting native bridge of SIP/2116-a406 and SIP/servidor-f5f7
Oct 7 18:23:36 WARNING[1875]: rtp.c:1418 ast_rtp_bridge: codec0 = 1024 is not codec1 = 2, cannot native bridge.
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-a406'
-- Executing Dial("SIP/2116-b4f3", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-7722 answered SIP/2116-b4f3
-- Attempting native bridge of SIP/2116-b4f3 and SIP/servidor-7722
Oct 7 18:24:26 NOTICE[1879]: rtp.c:516 ast_rtp_read: Unknown RTP codec 98 received
= Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-b4f3'
-- Executing Dial("SIP/2116-a289", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
= No one is available to answer at this time
Oct 7 18:25:23 WARNING[1883]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-dc81", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
= No one is available to answer at this time
Oct 7 18:26:06 WARNING[1890]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
```

Figura C19. Resultado de la prueba con el codec iLBC en la consola de Asterisk



```
Asterisk Console on 'G11T3' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda

-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
== Unregistered 'ODBC' CDR backend
== Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-0b0f", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct  7 18:26:56 WARNING[1915]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-fa7a", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct  7 18:27:46 WARNING[1919]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-1724", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-e2ce answered SIP/2116-1724
-- Attempting native bridge of SIP/2116-1724 and SIP/servidor-e2ce
Oct  7 18:28:00 WARNING[1923]: rtp.c:1418 ast_rtp_bridge: codec0 = 1024 is not codec1 = 8, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-1724'
-- Executing Dial("SIP/2116-d02f", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-fd94 answered SIP/2116-d02f
-- Attempting native bridge of SIP/2116-d02f and SIP/servidor-fd94
Oct  7 18:28:45 WARNING[1926]: rtp.c:1418 ast_rtp_bridge: codec0 = 1024 is not codec1 = 8, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-d02f'
```

Figura C20. Resultado de la prueba con el codec iLBC en la consola de Asterisk

En la figura C21 se puede observar al codec L16PCM Wideband del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del sipphone:

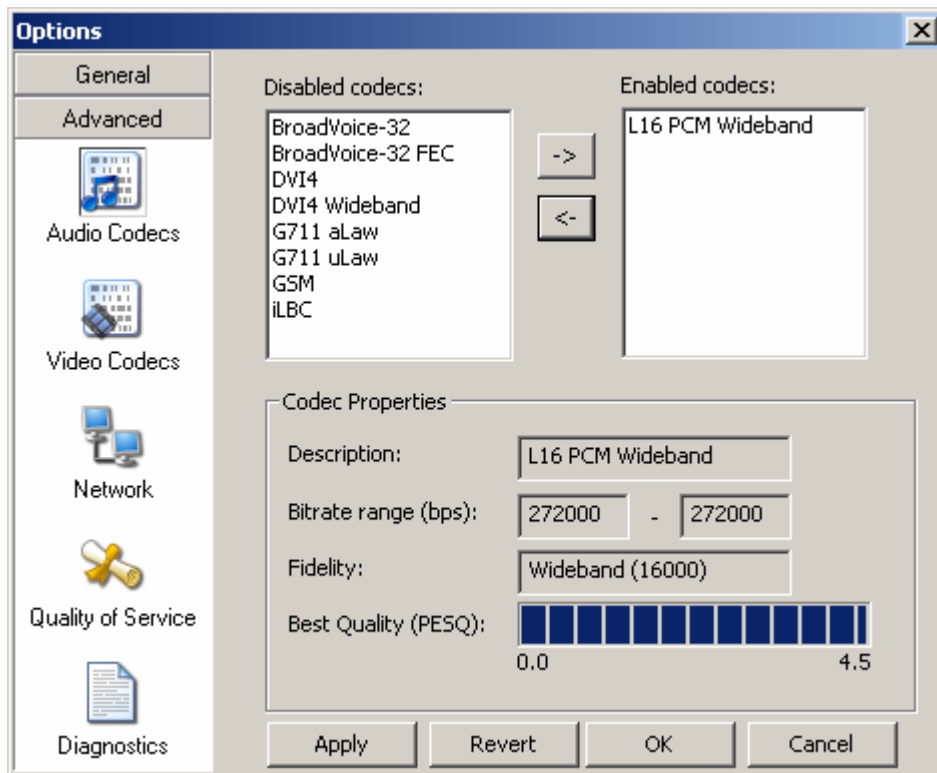
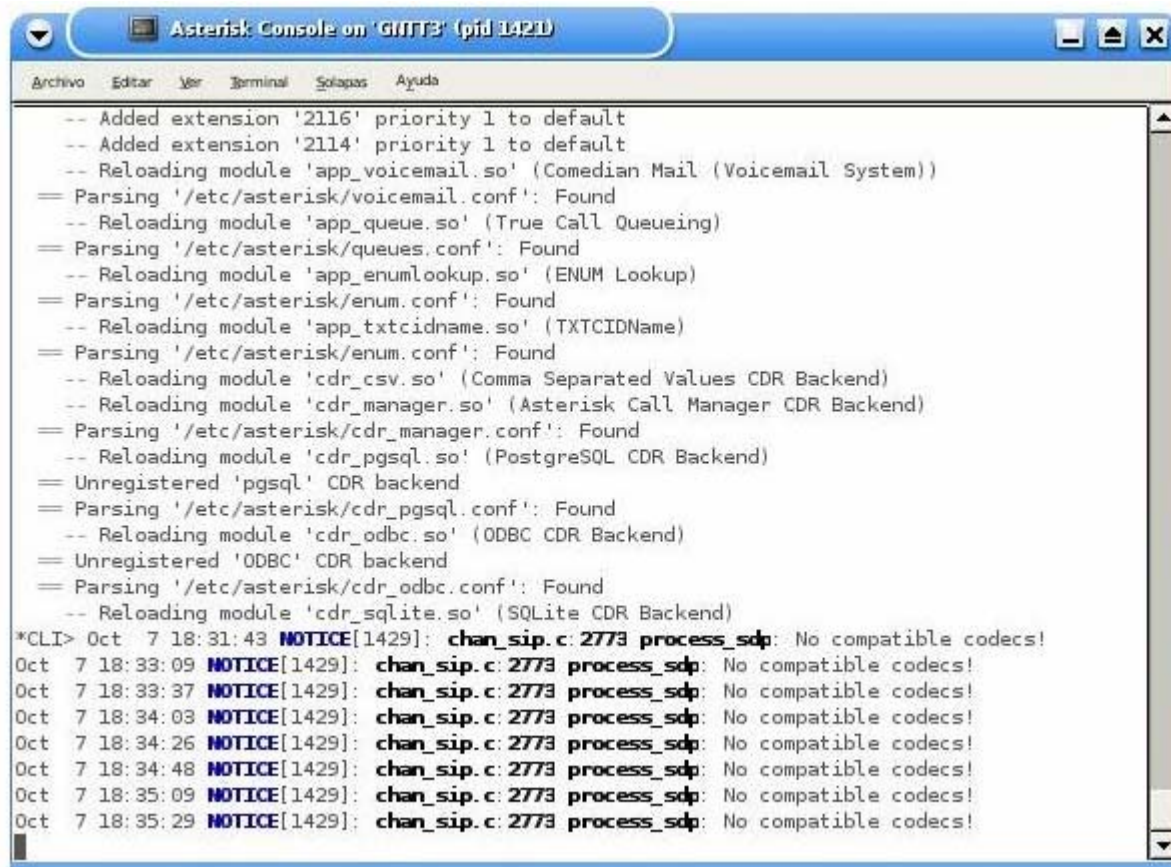


Figura C21. Softphone X-lite con solo el codec L16PCM Wideband habilitado.

- GSM 6.10 no compatibles
- iLBC-30ms no compatibles
- iLBC-20ms no compatibles
- Speex 15.2K no compatibles
- Speex 15.2K 40 ms no compatibles
- Speex 8.0K 40 ms no compatibles
- ITU G711 alaw no compatibles
- ITU G711 ulaw no compatibles

En la figura C22 se ve el resultado de la prueba en la consola de Asterisk.



```
Asterisk Console on 'G11T3' (pid 1421)
Archivo  Editar  Ver  Terminal  Solapas  Ayuda
-- Added extension '2116' priority 1 to default
-- Added extension '2114' priority 1 to default
-- Reloading module 'app_voicemail.so' (Comedian Mail (Voicemail System))
= Parsing '/etc/asterisk/voicemail.conf': Found
-- Reloading module 'app_queue.so' (True Call Queueing)
= Parsing '/etc/asterisk/queues.conf': Found
-- Reloading module 'app_enumlookup.so' (ENUM Lookup)
= Parsing '/etc/asterisk/enum.conf': Found
-- Reloading module 'app_txtcidname.so' (TXTCIDName)
= Parsing '/etc/asterisk/enum.conf': Found
-- Reloading module 'cdr_csv.so' (Comma Separated Values CDR Backend)
-- Reloading module 'cdr_manager.so' (Asterisk Call Manager CDR Backend)
= Parsing '/etc/asterisk/cdr_manager.conf': Found
-- Reloading module 'cdr_pgsq1.so' (PostgreSQL CDR Backend)
= Unregistered 'pgsq1' CDR backend
= Parsing '/etc/asterisk/cdr_pgsq1.conf': Found
-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
= Unregistered 'ODBC' CDR backend
= Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
*CLI> Oct  7 18:31:43 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:33:09 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:33:37 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:34:03 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:34:26 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:34:48 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:35:09 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
Oct  7 18:35:29 NOTICE[1429]: chan_sip.c: 2773 process_sdp: No compatible codecs!
```

Figura C22. Resultado de la prueba con el codec L16PCM Wideband en la consola de Asterisk

En la figura C23 se puede observar al codec ITU G711 alaw del softphone X-lite del cliente habilitado para probarlo con cada uno de los codecs del softphone:

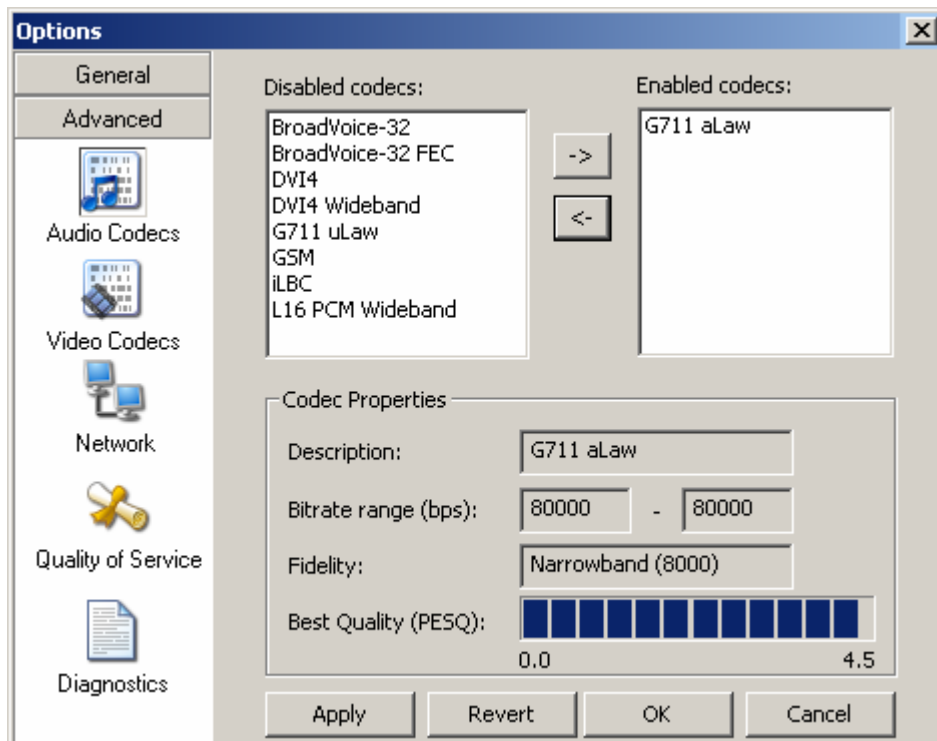
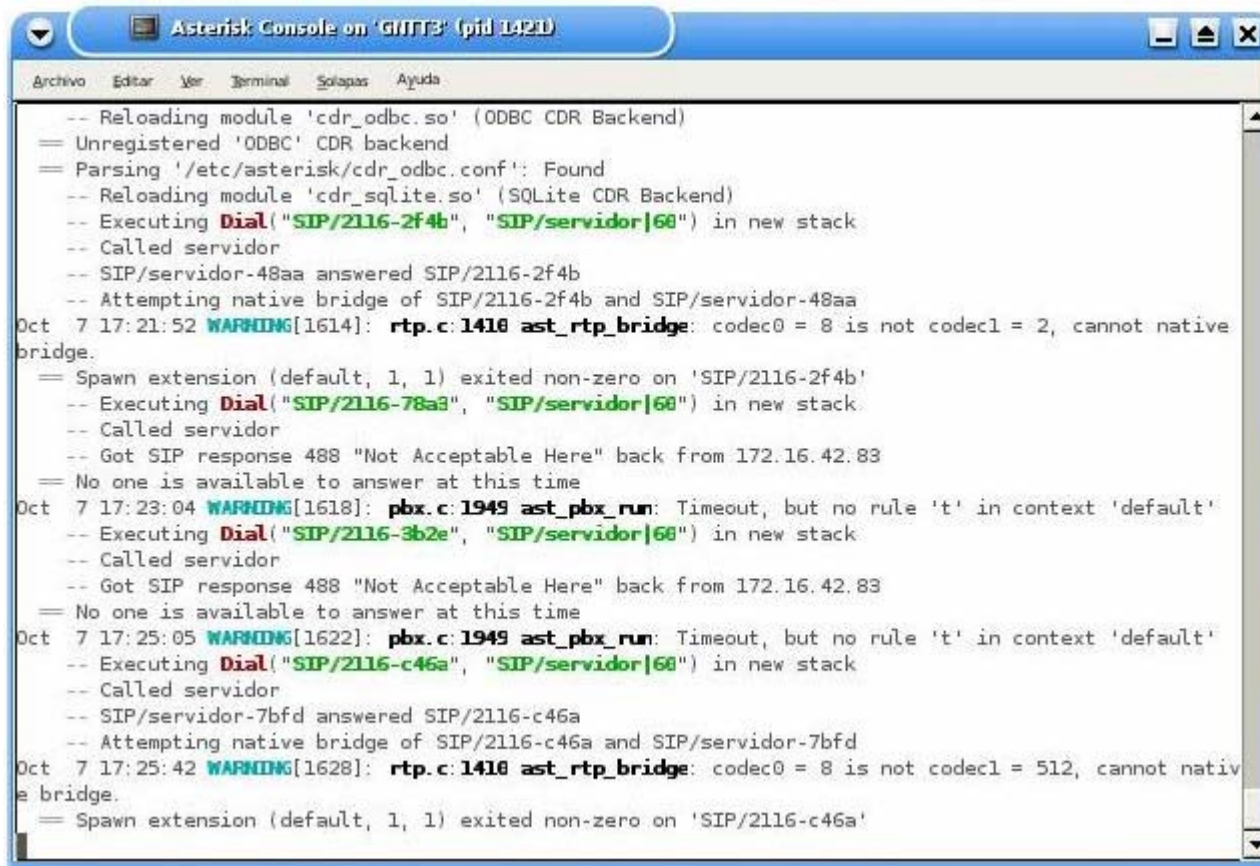


Figura C23. Softphone X-lite con solo el codec ITU G711 alaw habilitado.

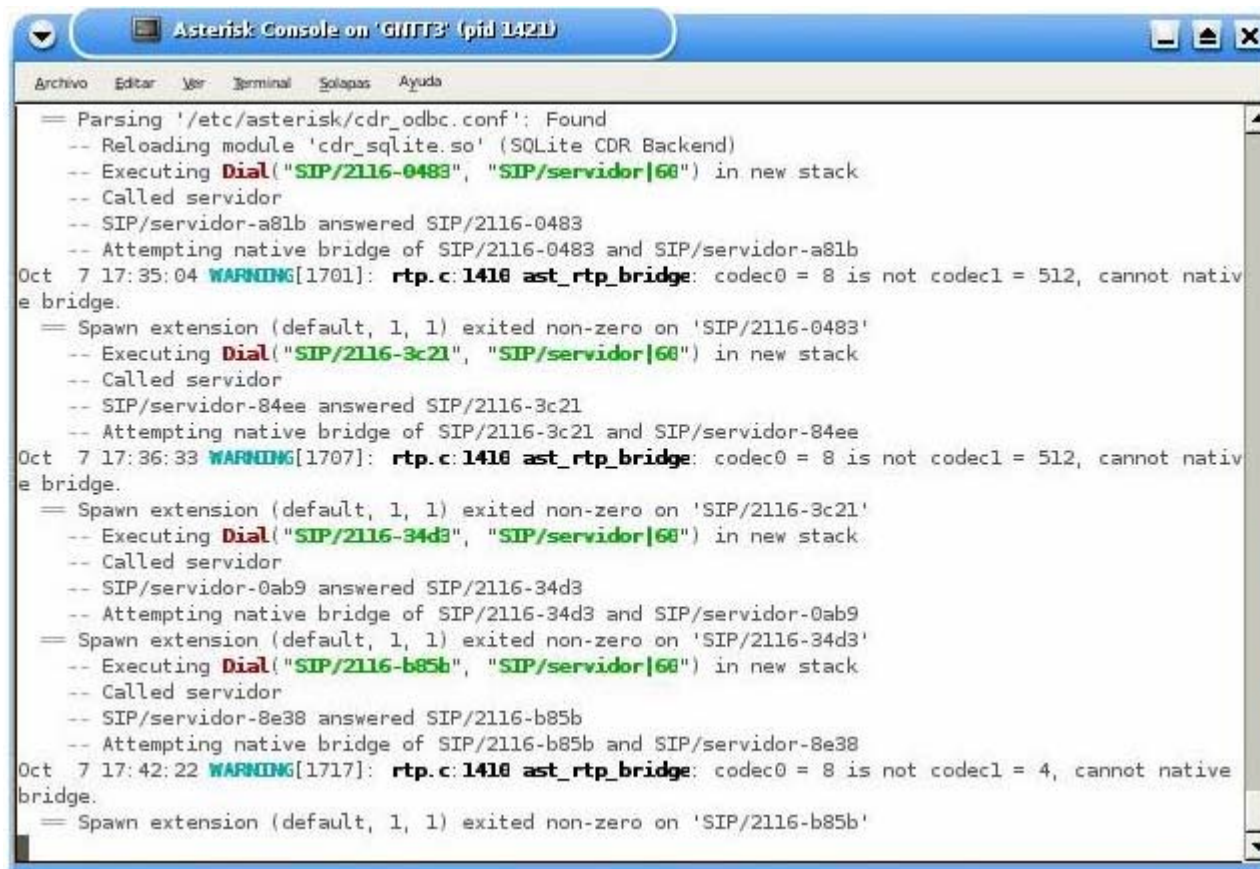
- GSM 6.10: La llamada se establece, la voz es entendible aunque se escucha a un volumen bajo y el retardo es significativo.
- iLBC-30ms: La llamada se genera pero no se establece.
- iLBC-20ms: La llamada se genera pero no se establece.
- Speex 15.2K: La llamada se establece, la voz es entendible aunque se escucha a un volumen bajo y el retardo es significativo.
- Speex 15.2K 40 ms: La llamada se establece, la voz es entendible aunque se escucha a un volumen bajo y el retardo es significativo pero menor que con el codec Speex 15.2K.
- Speex 8.0K 40 ms: La llamada se establece, la voz es entendible aunque se escucha a un volumen bajo y el retardo es significativo.
- ITU G711 alaw: La llamada se establece, la voz es clara, entendible pero se presenta un retardo no considerable.
- ITU G711 ulaw: La llamada se establece, la voz es clara, entendible y se presenta un retardo no considerable.

En las figuras C24 y C25 se ve el resultado de la prueba en la consola de Asterisk.



```
-- Reloading module 'cdr_odbc.so' (ODBC CDR Backend)
== Unregistered 'ODBC' CDR backend
== Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-2f4b", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-48aa answered SIP/2116-2f4b
-- Attempting native bridge of SIP/2116-2f4b and SIP/servidor-48aa
Oct 7 17:21:52 WARNING[1614]: rtp.c:1418 ast_rtp_bridge: codec0 = 8 is not codec1 = 2, cannot native
bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-2f4b'
-- Executing Dial("SIP/2116-78a3", "SIP/servidor|60") in new stack.
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 17:23:04 WARNING[1618]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-3b2e", "SIP/servidor|60") in new stack
-- Called servidor
-- Got SIP response 488 "Not Acceptable Here" back from 172.16.42.83
== No one is available to answer at this time
Oct 7 17:25:05 WARNING[1622]: pbx.c:1949 ast_pbx_run: Timeout, but no rule 't' in context 'default'
-- Executing Dial("SIP/2116-c46a", "SIP/servidor|60") in new stack
-- Called servidor
-- SIP/servidor-7bfd answered SIP/2116-c46a
-- Attempting native bridge of SIP/2116-c46a and SIP/servidor-7bfd
Oct 7 17:25:42 WARNING[1628]: rtp.c:1418 ast_rtp_bridge: codec0 = 8 is not codec1 = 512, cannot nativ
e bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-c46a'
```

Figura C24. Resultado de la prueba con el codec ITU G711 alaw



```
Asterisk Console on 'G11T3' (pid 1421)
Archivo  Editor  Ver  Terminal  Solapas  Ayuda

== Parsing '/etc/asterisk/cdr_odbc.conf': Found
-- Reloading module 'cdr_sqlite.so' (SQLite CDR Backend)
-- Executing Dial("SIP/2116-0483", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-a81b answered SIP/2116-0483
-- Attempting native bridge of SIP/2116-0483 and SIP/servidor-a81b
Oct 7 17:35:04 WARNING[1701]: rtp.c:1418 ast_rtp_bridge: codec0 = 8 is not codec1 = 512, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-0483'
-- Executing Dial("SIP/2116-3c21", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-84ee answered SIP/2116-3c21
-- Attempting native bridge of SIP/2116-3c21 and SIP/servidor-84ee
Oct 7 17:36:33 WARNING[1707]: rtp.c:1418 ast_rtp_bridge: codec0 = 8 is not codec1 = 512, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-3c21'
-- Executing Dial("SIP/2116-34d3", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-0ab9 answered SIP/2116-34d3
-- Attempting native bridge of SIP/2116-34d3 and SIP/servidor-0ab9
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-34d3'
-- Executing Dial("SIP/2116-b85b", "SIP/servidor[60]") in new stack
-- Called servidor
-- SIP/servidor-8e38 answered SIP/2116-b85b
-- Attempting native bridge of SIP/2116-b85b and SIP/servidor-8e38
Oct 7 17:42:22 WARNING[1717]: rtp.c:1418 ast_rtp_bridge: codec0 = 8 is not codec1 = 4, cannot native bridge.
== Spawn extension (default, 1, 1) exited non-zero on 'SIP/2116-b85b'
```

Figura C25. Resultado de la prueba con el codec ITU G711 alaw