Event List

0. About

All the events that can be handled by FreeSWITCH's event system.

- The current up-to-date list of events can be found in `src/switch_event.c` in a char array called `EVENT_NAMES`.

1. Channel events

The channel events are event types/classes that can be used to monitor which calls come into an extension, and what states are the calls currently in.

Channel events can carry additional information for the protocol (i.e. SIP) used to transport the call. This information can safely be ignored, you don't have to use it to be able to track calls.

The minimum information sent for channel events (plus the headers in 0.1 Minimum event information):

```plaintext
Channel-State: CS_NEW
Channel-State-Number: 0
Channel-Name: sofia/192.168.0.58/jonas@192.168.0.58%3A5060
```

- Channel states

  The state the channel can be in. Indicated by the Channel-State and Channel-State-Number, which are sent with all CHANNEL events.

  See Channel States for the complete list.

1.1 CHANNEL_CALLSTATE event

- TODO
There seems to be a correspondence between channel states and `CHANNEL_*STATE` events. For example, only saw `CHANNEL_CALLSTATE` with channel states `RINGING` and `HANGUP`, and `CHANNEL_STATE` with the rest.

Also corroborated by a mailing list thread: https://lists.freeswitch.org/pipermail/freeswitch-users/2012-February/080189.html, and the answer may lies in `switch_core_state_machine.c`.

Creating a New Endpoint: Lifecycle of a Session page also goes into the FreeSWITCH state machine.
The empty lines denote the boundaries between different events. Made a test call (that hasn't been answered, only routed by an extension), and the resulting events have been filtered in Vim using `/"Channel-State"\|^"Channel-Call-State"|CHANNEL_CALLSTATE|CHANNEL_STATE`

The UML diagram in Life Cycle of a Call and the dump below should be helpful to decipher this. See attachment for a diff between two CHANNEL_STATE event, one in CS_ROUTING and one in CS_EXECUTE state (there's very little difference).

See also Creating a New Endpoint: Lifecycle of a Session page!
"Channel-State","CS_NEW"},
"Channel-State","CS_INIT"},

{"Channel-Call-State","DOWN"},
{"Channel-State","CS_NEW"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","DOWN"},
{"Channel-State","CS_INIT"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","DOWN"},
{"Channel-State","CS_INIT"},
{"Event-Name","CHANNEL_CREATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_ROUTING"},
{"Event-Name","CHANNEL_CALLSTATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_ROUTING"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_EXECUTE"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_EXECUTE"},
{"Event-Name","CHANNEL_EXECUTE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_EXECUTE"},
{"Event-Name","CHANNEL_EXECUTE_COMPLETE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_ROUTING"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_ROUTING"},
{"Event-Name","CHANNEL_HANGUP"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_HANGUP"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","RINGING"},
{"Channel-State","CS_HANGUP"},
{"Event-Name","CHANNEL_CALLSTATE"},

{"Channel-Call-State","HANGUP"},
{"Channel-State","CS_HANGUP"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","HANGUP"},
{"Channel-State","CS_REPORTING"},
{"Event-Name","CHANNEL_STATE"},

{"Channel-Call-State","HANGUP"},
{"Channel-State","CS_REPORTING"},
{"Event-Name","CHANNEL_HANGUP_COMPLETE"},

{"Channel-Call-State","HANGUP"},
{"Channel-State","CS_REPORTING"},
{"Event-Name","CHANNEL_DESTROY"},

{"Channel-Call-State","HANGUP"},
{"Channel-State","CS_DESTROY"},
{"Event-Name","CHANNEL_STATE"},
1.2 CHANNEL_CREATE event

CHANNEL_CREATE is sent when an extension is going to do something. It can either be dialing someone or it can be an incoming call to an extension.

This event does not have any additional information.
### CHANNEL_CREATE example

- **Event-Name:** CHANNEL_CREATE
- **Core-UUID:** 17c1c070-8a13-11de-9ab6-91a5c9f91e77
- **FreeSWITCH-Hostname:** jmesquita-dell
- **FreeSWITCH-IPv4:** 186.18.21.203
- **FreeSWITCH-IPv6:** %3A%3A1
- **Event-Date-Local:** 2009-08-16 00:26:08
- **Event-Date-GMT:** Sun, 16 Aug 2009 03:26:08 GMT
- **Event-Date-Timestamp:** 1250393168131798
- **Event-Calling-File:** switch_channel.c
- **Event-Calling-Function:** switch_channel_set_caller_profile
- **Event-Calling-Line-Number:** 1428
- **Channel-State:** CS_NEW
- **Channel-State-Number:** 0
- **Channel-Name:** sofia/internal/1000@192.168.1.100
- **Unique-ID:** 89aaa4da-8a14-11de-9ab6-91a5c9f91e77
- **Call-Direction:** inbound
- **Presence-Call-Direction:** inbound
- **Answer-State:** ringing

### 1.3 CHANNEL_DESTROY event

Called when a channel should get destroyed.

### CHANNEL_DESTROY example

- **Channel-Read-Codec-Name:** PCMU
- **Channel-Read-Codec-Rate:** 8000
- **Channel-Write-Codec-Name:** PCMU
- **Channel-Write-Codec-Rate:** 8000
- **Caller-Username:** jonas
- **Caller-Dialplan:** XML
- **Caller-Caller-ID-Name:** jonas
- **Caller-Caller-ID-Number:** jonas
- **Caller-Network-Addr:** 192.168.0.58
- **Caller-Destination-Number:** 192.168.0.58/arne%192.168.0.58
- **Caller-Unique-ID:** f66e8e31-c9fb-9b41-a9a2-a1586facb97f
- **Caller-Source:** mod_sofia
- **Caller-Context:** default
- **Caller-Channel-Name:** sofia/192.168.0.58/arne
- **Caller-Screen-Bit:** yes
- **Caller-Privacy-Hide-Name:** no
- **Caller-Privacy-Hide-Number:** no
- **Originator-Username:** jonas
- **Originator-Dialplan:** XML
- **Originator-Caller-ID-Name:** jonas
- **Originator-Caller-ID-Number:** jonas
- **Originator-Network-Addr:** 192.168.0.58
- **Originator-Destination-Number:** 541
- **Originator-Unique-ID:** 3dd4ef4f7-36ed-a04d-a8f7-7aebb683af50
- **Originator-Source:** mod_sofia
- **Originator-Context:** default
- **Originator-Channel-Name:** sofia/192.168.0.58/jonas%40192.168.0.58%3A5060
- **Originator-Screen-Bit:** yes
- **Originator-Privacy-Hide-Name:** no
- **Originator-Privacy-Hide-Number:** no

### 1.4 CHANNEL_STATE event

Sent when a channel has switched its calls state.

This event does not contain any additional information.
1.5 CHANNEL_ANSWER event

Someone calls and answer
Channel-Read-Codec-Rate:  8000
Channel-State:  CS_CONSUME_MEDIA
Channel-State-Number:  7
Channel-Write-Codec-Bit-Rate:  64000
Channel-Write-Codec-Name:  PCMU
Channel-Write-Codec-Rate:  8000
Core-UUID:  347c8e76-2a34-423a-8199-50860933a276
Event-Calling-File:  switch_channel.c
Event-Calling-Function:  switch_channel_perform_mark_answered
Event-Calling-Line-Number:  3397
Event-Date-GMT:  Sun, %2013%20Jan%202013%2015%3A38%3A48%20GMT
Event-Date-Local:  2013-01-13%2010%3A38%3A48
Event-Date-Timestamp:  135809152873446
Event-Name:  CHANNEL_ANSWER
Event-Sequence:  6726
FreeSWITCH-Hostname:  aztrock-home
FreeSWITCH-IPv4:  192.168.1.11
FreeSWITCH-IPv6:  %3A%3A1
FreeSWITCH-Switchname:  aztrock-home
Other-Leg-ANI:  100
Other-Leg-Caller-ID-Name:  100
Other-Leg-Caller-ID-Number:  100
Other-Leg-Channel-Answered-Time:  0
Other-Leg-Channel-Bridged-Time:  0
Other-Leg-Channel-Created-Time:  0
Other-Leg-Channel-Hangup-Time:  0
Other-Leg-Channel-Hold-Accum:  0
Other-Leg-Channel-Last-Hold:  0
Other-Leg-Channel-Name:  sofia/internal/100%40192.168.1.11
Other-Leg-Channel-Progress-Media-Time:  0
Other-Leg-Channel-Progress-Time:  135809152743428
Other-Leg-Channel-Resurrect-Time:  0
Other-Leg-Channel-Transfer-Time:  0
Other-Leg-Context:  default
Other-Leg-Destination-Number:  1000
Other-Leg-Dialplan:  XML
Other-Leg-Direction:  inbound
Other-Leg-Network-Addr:  192.168.1.11
Other-Leg-Privacy-Hide-Name:  false
Other-Leg-Privacy-Hide-Number:  false
Other-Leg-Profile-Created-Time:  0
Other-Leg-Source:  mod_sofia
Other-Leg-Unique-ID:  c9bbde8b-379b-45d4-b193-3f761a44f3e2
Other-Leg-Username:  100
Other-Type:  originator
Presence-Call-Direction:  outbound
Unique-ID:  81273088-c31f-4469-85a6-c878e42210e5
variable_RFC2822_DATE:  Sun, %2013%20Jan%202013%2010%3A38%3A47%20-0500
variable_absolute_codec_string:  PCMU%4080000h%4020i%4064000b,PCMA%4080000h%4020i%4064000b
variable_advertised_media_ip:  192.168.1.11
variable_call_uuid:  c9bbde8b-379b-45d4-b193-3f761a44f3e2
variable_channel_name:  sofia/internal/sip%3A1000%40192.168.1.11%3A5062
variable_dialed_domain:  192.168.1.11
variable_dialed_extension:  1000
variable_dialed_user:  1000
variable_dtmf_type:  info
variable_export_vars:  RFC2822_DATE,dialed_extension
variable_is_outbound:  true
variable_local_media_ip:  192.168.1.11
variable_local_media_port:  20342
variable_max_forwards:  69
variable_read_codec:  PCMU
variable_read_rate:  8000
variable_recovery_profile_name: internal
variable_remote_media_ip: 192.168.1.11
variable_remote_media_port: 30882
variable_rtp_use_ssrc: 3506474416
variable_session_id: 66
variable_signal_bond: c9bbde8b-379b-45d4-b193-3f761a44f3e2
variable_sip_audio_recv_pt: 0
variable_sip_call_id: 29869441-d83a-1230-60eb69774d98
variable_sip_contact_host: 192.168.1.11
variable_sip_contact_port: 5062
variable_sip_contact_uri: 1000%40192.168.1.11%3A5062
variable_sip_contact_user: 1000
variable_sip_cseq: 38709699
variable_sip_destination_url: sip%3A1000%40192.168.1.11%3A5062
variable_sip_from_display: Extension%20100
variable_sip_from_host: 192.168.1.11
variable_sip_from_tag: c9bbde8b-379b-45d4-b193-3f761a44f3e2
variable_sip_from_uri: 1000%40192.168.1.11%3A5062
variable_sip_full_from: %22Extension%20100%22%3Bsip%3A1000%40192.168.1.11%3A5062%3Btag%3Db14H3mpvR9N5m
variable_sip_full_to: %3C%3Cip%3A1000%40192.168.1.11%3A5062%3E%3Btag%3D90972448-fb32-4a25-b753-2c6a56174df1
variable_sip_full_via: SIP/2.0/UDP%20192.168.1.11%3Bport%3D5060%3Breceived%3D192.168.1.11%3Bbranch%3D3D9h04bi74m9nup22bej
variable_sip_invite_domain: 192.168.1.11
variable_sip_local_network_addr: 181.133.83.254
variable_sip_local_sdp_str: v%3D0%0Ao%3DFreeSWITCH%201358071185%201358071186%20IN%20IP4%20192.168.1.11%0Aa%3DFreeSWITCH%0Ac%3DIN%20IP4%20192.168.1.11%0At%3D0%200%0Am%3Daudio%2020342%20RTP/AVP%20200%20201%201%2013%0Aa%3Drtpmap%3A101%20telephone-event/8000%0Aa%3Dsendrecv%0A
variable_sip_network_ip: 192.168.1.11
variable_sip_network_port: 5062
variable_sip_outgoing_contact_uri: %3Csip%3Amod_sofia%40192.168.1.11%3A5062%3E
variable_sip_profile_name: internal
variable_sip_recover_contact: %3Csip%3A1000%40192.168.1.11%3A5062%3E
variable_sip_recover_via: SIP/2.0/UDP%20192.168.1.11%3Bport%3D5060%3Breceived%3D192.168.1.11%3Bbranch%3D3D9h04bi74m9nup22bej
variable_sip_reply_host: 192.168.1.11
variable_sip_reply_port: 5062
variable_sip_req_uri: 1000%40192.168.1.11%3A5062
variable_sip_to_host: 192.168.1.11
variable_sip_to_tag: 90972448-fb32-4a25-b753-2c6a56174df1
variable_sip_to_uri: 1000%40192.168.1.11%3A5062
variable_sip_to_user: 1000
variable_sip_use_codec_name: PCMU
variable_sip_use_codec_ptime: 20
variable_sip_use_codec_rate: 8000
variable_sip_use_pt: 0
variable_sofia_profile_name: internal
variable_switch_m_sdp: v%3D0%0Ao%3Daztrock-home%203567080327%200%20IN%20IP4%20192.168.1.11%0Aa%3Daztrock-home%0Ac%3DIN%20IP4%20192.168.1.11%0At%3D0%200%0Am%3Daudio%2030882%20RTP/AVP%20200%20201%201%2013%0Aa%3Drtpmap%3A0%20PCMU
variable_switch_r_sdp: v%3D0%0Ao%3Daztrock-home%203567080327%200%20IN%20IP4%20192.168.1.11%0Aa%3Daztrock-home%0Ac%3DIN%20IP4%20192.168.1.11%0At%3D0%200%0Am%3Daudio%2030882%20RTP/AVP%20200%20201%201%2013%0Aa%3Drtpmap%3A0%20PCMU
variable_uuid: 81273088-c31f-4469-85a6-c878e42210e5
variable_write_codec: PCMU
variable_write_rate: 8000

CHANNEL_ANSWER - inbound example

Answer-State: answered
Call-Direction: inbound
Caller-ANI: 100
Caller-Callee-ID-Name: Outbound%20Call
Caller-Callee-ID-Number: 1000
null
1.6 CHANNEL_HANGUP event

One of the users has hangup.

TODO How do we know which one?

CHANNEL_HANGUP example

Hangup-Cause: NORMAL_CLEARING
Channel-Read-Codec-Name: PCMU
Channel-Read-Codec-Rate: 8000
Channel-Write-Codec-Name: PCMU
Channel-Write-Codec-Rate: 8000
Caller-Username: jonas
Caller-Dialplan: XML
Caller-Caller-ID-Name: jonas
Caller-Caller-ID-Number: jonas
Caller-Network-Addr: 192.168.0.58
Caller-Destination-Number: 541
Caller-Unique-ID: 0dd4e4f7-36ed-a04d-a8f7-7ae683af50
Caller-Source: mod_sofia
Caller-Context: default
Caller-Channel-Name: sofia/192.168.0.58/jonas%40192.168.0.58%3A5060
Caller-Screen-Bit: yes
Caller-Privacy-Hide-Name: no
Caller-Privacy-Hide-Number: no
Originatee-Username: jonas
Originatee-Dialplan: XML
Originatee-Caller-ID-Name: jonas
Originatee-Caller-ID-Number: jonas
Originatee-Network-Addr: 192.168.0.58
Originatee-Destination-Number: 192.168.0.58/arne%25192.168.0.58
Originatee-Unique-ID: f66e8e31-c9fb-9b41-a9a2-a1586facb97f
Originatee-Source: mod_sofia
Originatee-Context: default
Originatee-Channel-Name: sofia/192.168.0.58/arne
Originatee-Screen-Bit: yes
Originatee-Privacy-Hide-Name: no
Originatee-Privacy-Hide-Number: no

Read more about possible hangup causes.

1.7 CHANNEL_HANGUP_COMPLETE event

CHANNEL_HANGUP_COMPLETE example

RECV EVENT
Event-Name: CHANNEL_HANGUP_COMPLETE
Core-UUID: 9b0de0b8-f55e-40d8-a2bd-17931ob53493
FreeSWITCH-Hostname: myhost
FreeSWITCH-IPv4: 192.168.0.2
FreeSWITCH-IPv6: ::1
Event-Date-Local: 2009-10-09 20:08:26
Event-Date-GMT: Sat, 10 Oct 2009 00:08:26 GMT
Event-Date-Timestamp: 125513306952270
Event-Calling-File: switch_core_state_machine.c
Event-Calling-Function: switch_core_session_hangup_state
Event-Calling-Line-Number: 503
Hangup-Cause: NORMAL_CLEARING
Channel-State: CS_HANGUP
Channel-State-Number: 10
Channel-Name: sofia/internal/1000@192.168.0.2
Unique-ID: e5a82e39-6dc1-4d7d-a300-aa9cd4284073
Call-Direction: inbound
Presence-Call-Direction: inbound
Answer-State: answered
Channel-Read-Codec-Name: GSM
Channel-Read-Codec-Rate: 8000
Channel-Write-Codec-Name: GSM
Channel-Write-Codec-Rate: 8000
Caller-Username: 1000
Caller-Dialplan: XML
Caller-Caller-ID-Name: 1000
Caller-Caller-ID-Number: 1000
Caller-Network-Addr: 192.168.0.104
Caller-Destination-Number: 3030
Caller-Unique-ID: e5a82e39-6dc1-4d7d-a300-aa9cd4284073
Caller-Source: mod_sofia
Caller-Context: default
Caller-Channel-Name: sofia/internal/1000@192.168.0.2
Caller-Profile-Index: 1
Caller-Profile-Created-Time: 1255133286498223
Caller-Channel-Answered-Time: 1255133286504829
Caller-Channel-Progress-Time: 0
Caller-Channel-Progress-Media-Time: 0
Caller-Channel-Hangup-Time: 1255133306952270
Caller-Channel-Transfer-Time: 0
Caller-Screen-Bit: true
Caller-Privacy-Hide-Name: false
Caller-Privacy-Hide-Number: false
variable_sip_received_ip: 192.168.0.104
variable_sip_received_port: 5060
variable_sip_via_protocol: udp
variable_sip_authorized: true
variable_sip_number_alias: 1000
variable_sip_auth_username: 1000
variable_sip_auth_realm: 192.168.0.2
variable_number_alias: 1000
variable_user_name: 1000
variable_domain_name: 192.168.0.2
variable_toll_allow: domestic,international,local
variable_accountcode: 1000
variable_user_context: default
variable_effective_caller_id_name: Extension 1000
variable_effective_caller_id_number: 1000
variable_outbound_caller_id_name: FreeSWITCH
variable_outbound_caller_id_number: 0000000000
variable_callgroup: techsupport
variable_record_stereo: true
variable_default_gateway: example.com
variable_default_areacode: 918
variable_transfer_fallback_extension: operator
variable_sip_from_params: transport=UDP
variable_sip_from_user: 1000
variable_sip_from_uri: 1000@192.168.0.2
variable_sip_from_host: 192.168.0.2
variable_sip_from_user_stripped: 1000
variable_sip_from_tag: 7bae8202
variable_sofia_profile_name: internal
variable_sip_req_params: transport=UDP
variable_sip_req_user: 1000
variable_sip_req_uri: 3030@192.168.0.2
variable_sip_req_host: 192.168.0.2
variable_sip_other_params: transport=UDP
variable_sip_contact_params: transport=UDP
variable_sip_contact_user: 1000
variable_sip_contact_port: 60780
variable_answermsec: 6
variable_progress_mediamsec: 0
variable_flow_billmsec: 20454
variable_uduration: 20454047
variable_billusec: 20447441
variable_progressusec: 0
variable_answerusec: 6606
variable_progress_mediausec: 0
variable_flow_billusec: 20454047
variable_rtp_audio_in_raw_bytes: 45765
variable_rtp_audio_in_media_bytes: 45630
variable_rtp_audio_in_packet_count: 1017
variable_rtp_audio_in_media_packet_count: 1014
variable_rtp_audio_in_skip_packet_count: 8
variable_rtp_audio_in_jb_packet_count: 0
variable_rtp_audio_in_dtmf_packet_count: 0
variable_rtp_audio_in_cng_packet_count: 0
variable_rtp_audio_in_flush_packet_count: 0
variable_rtp_audio_out_raw_bytes: 44055
variable_rtp_audio_out_media_bytes: 44055
variable_rtp_audio_out_packet_count: 979
variable_rtp_audio_out_media_packet_count: 979
variable_rtp_audio_out_skip_packet_count: 0
variable_rtp_audio_out_dtmf_packet_count: 0
variable_rtp_audio_out_cng_packet_count: 0

1.8 CHANNEL_EXECUTE event

This event indicates that the PBX is doing something with the call. (Typically looking in the dial plan).
Found usage examples in IVR using mod_erlang_event, but couldn't find the Application* headers documented anywhere. mod_event_socket's sendmsg section only mentions the Application-UUID header:

If you would like to correlate the CHANNEL_EXECUTE and CHANNEL_EXECUTE_COMPLETE events that are generated when the command you send using sendmsg is executed you can add an Event-UUID header with a UUID you specify. In the corresponding events, the UUID will be in the Application-UUID header. If you do not specify an Event-UUID, Freeswitch will generate a UUID for the Application-UUID.

Example:

Event-UUID: 22075ce5-b67b-4f04-a6dd-1726ec14c8bf

The Application* headers seem to be tightly coupled with CHANNEL_EXECUTE and CHANNEL_EXECUTE_COMPLETE headers:

https://github.com/signalwire/freeswitch/blob/15ad4c23e259c1c2df5f8f89e9124e5f36d2e94/src/switch_core_session.c#L2870

```c
if (switch_event_create(&event, SWITCH_EVENT_CHANNEL_EXECUTE) == SWITCH_STATUS_SUCCESS) {
    switch_channel_event_set_data(session->channel, event);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application", application_interface->interface_name);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-Data", expanded);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-UUID", app_uuid);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-UUID-Name", app_uuid_name);
    switch_event_fire(&event);
}
```

https://github.com/signalwire/freeswitch/blob/15ad4c23e259c1c2df5f8f89e9124e5f36d2e94/src/switch_core_session.c#L2893

```c
if (switch_event_create(&event, SWITCH_EVENT_CHANNEL_EXECUTE_COMPLETE) == SWITCH_STATUS_SUCCESS) {
    const char *resp = switch_channel_get_variable(session->channel, SWITCH_CURRENT_APPLICATION_RESPONSE_VARIABLE);
    switch_event_set_data(session->channel, event);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application", application_interface->interface_name);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-Data", expanded);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-Response", resp ? resp : "_none_"),
    app_uuid);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-UUID", app_uuid);
    switch_event_add_header_string(event, SWITCH_STACK_BOTTOM, "Application-UUID-Name", app_uuid_name);
    switch_event_fire(&event);
}
```
### CHANNEL_EXECUTE example

Channel-Read-Codec-Name: PCMU  
Channel-Read-Codec-Rate: 8000  
Channel-Write-Codec-Name: PCMU  
Channel-Write-Codec-Rate: 8000  
Caller-Username: jonas  
Caller-Dialplan: XML  
Caller-Caller-ID-Name: jonas  
Caller-Caller-ID-Number: jonas  
Caller-Network-Addr: 192.168.0.58  
Caller-Destination-Number: 541  
Caller-Unique-ID: 0dd4e4f7-36ed-a04d-a8f7-7aebb683af50  
Caller-Source: mod_sofia  
Caller-Context: default  
Caller-Channel-Name: sofia/192.168.0.58/jonas%40192.168.0.58%3A5060  
Caller-Privacy-Hide-Name: no  
Caller-Privacy-Hide-Number: no

#### 1.9 CHANNEL_EXECUTE_COMPLETE event

This event is emitted when a dialplan application (started with `sendmsg` or `send (?)`) finished running. 

```  
Todo: See TODO at section 1.8 CHANNEL_EXECUTE event
```

### CHANNEL_EXECUTE_COMPLETE example

RECV EVENT  
Event-Name: CHANNEL_EXECUTE_COMPLETE  
Core-UUID: 9b0de0b8-f55e-40d8-a2bd-179310b53493  
FreeSWITCH-Hostname: myhost  
FreeSWITCH-IPv4: 192.168.0.2  
FreeSWITCH-IPv6: ::1  
Event-Date-Local: 2009-10-09 20:08:26  
Event-Date-GMT: Sat, 10 Oct 2009 00:08:26 GMT  
Event-Date-Timestamp: 1255133306952270  
Event-Calling-File: switch_core_session.c  
Event-Calling-Function: switch_core_session_exec  
Event-Calling-Line-Number: 1480  
Channel-State: CS_HANGUP  
Channel-State-Number: 10  
Channel-Name: sofia/internal/1000@192.168.0.2  
Unique-ID: e5a82e39-6dcl-4d7d-a300-aa9cd4284073  
Call-Direction: inbound  
Presence-Call-Direction: inbound  
Answer-State: answered  
Channel-Read-Codec-Name: GSM  
Channel-Read-Codec-Rate: 8000  
Channel-Write-Codec-Name: GSM  
Channel-Write-Codec-Rate: 8000  
Caller-Username: 1000  
Caller-Dialplan: XML  
Caller-Caller-ID-Name: 1000  
Caller-Caller-ID-Number: 1000  
Caller-Network-Addr: 192.168.0.104  
Caller-Destination-Number: 3030  
Caller-Unique-ID: e5a82e39-6dcl-4d7d-a300-aa9cd4284073  
Caller-Source: mod_sofia  
Caller-Context: default  
Caller-Channel-Name: sofia/192.168.0.58/1000%40192.168.0.58%3A5060  
Caller-Profile-Index: 1  
Caller-Profile-Created-Time: 1255133286498223  
Caller-Channel-Created-Time: 1255133286498223  
Caller-Channel-Answered-Time: 1255133286504829  
Caller-Channel-Progress-Time: 0
Caller-Channel-Progress-Media-Time: 0
Caller-Channel-Hangup-Time: 0
Caller-Channel-Transfer-Time: 0
Caller-Screen-Bit: true
Caller-Privacy-Hide-Name: false
Caller-Privacy-Hide-Number: false
variable_sip_received_ip: 192.168.0.104
variable_sip_received_port: 5060
variable_sip_via_protocol: udp
variable_sip_authorized: true
variable_sip_number_alias: 1000
variable_sip_auth_username: 1000
variable_number_alias: 1000
variable_user_name: 1000
variable_domain_name: 192.168.0.2
variable_domain_name: 192.168.0.2
variable_toll_allow: domestic,international,local
variable_accountcode: 1000
variable_user_context: default
variable_effective_caller_id_name: Extension 1000
variable_effective_caller_id_number: 1000
variable_outbound_caller_id_name: FreeSWITCH
variable_outbound_caller_id_number: 0000000000
variable_callgroup: techsupport
variable_record_stereo: true
variable_default_gateway: example.com
variable_default_area_code: 918
variable_transfer_fallback_extension: operator
variable_sip_from_params: transport=UDP
variable_sip_from_user: 1000
variable_sip_from_uri: 1000@192.168.0.2
variable_sip_from_host: 192.168.0.2
variable_sip_from_stripped: 1000
variable_sip_from_tag: 7bae8202
variable_sip_via_protocol: internal
variable_sip_auth_username: 1000
variable_sip_auth_realm: 192.168.0.2
variable_sip_contact_params: transport=UDP
variable_sip_contact_user: 1000
variable_sip_contact_port: 60780
variable_sip_contact_uri: 1000@190.52.138.225:60780
variable_sip_contact_host: 190.52.138.225
variable_channel_name: sofia/internal/1000@192.168.0.2
variable_sip_call_id: ODZhNDk5YzlmZDg3YTExOWU3NmM2MzI1MzFmNDU.
variable_sip_via_host: 190.52.138.225
variable_sip_via_port: 60780
variable_sip_via_rport: 5060
variable_max_forwards: 70
variable_presence_id: 1000@192.168.0.2
variable_switch_s_adp: v=0
a=2 0 0 IN IP4 190.52.138.225
a=2 c=IN IP4 190.52.138.225
m=audio 60790 RTP/AVP 3 110 98 8 0 101
a=rtpmap:3 GSM/8000
a=rtpmap:110 speex/8000
a=rtpmap:98 iLBC/8000
a=fmtp:98 mode=30
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
variable_remote_media_ip: 190.52.138.225
variable_remote_media_port: 60790
variable_write_codec: GSM
variable_write_rate: 8000
variable_local_media_ip: 192.168.0.2
variable_local_media_port: 19878
variable_endpoint_disposition: ANSWER
variable_current_application_data: $1-192.168.0.2@default
variable_current_application: conference
variable_sip_term_status: 200
variable_sip_term_cause: 16
variable_sip_hangup_disposition: recv_bye
variable_read_codec: GSM
variable_read_rate: 8000
Application: conference
Application-Data: $1-192.168.0.2@default
Application-Response: _none

1.10 CHANNEL_BRIDGE event

A call is being bridged between two endpoints.

**CHANNEL_BRIDGE example**

<table>
<thead>
<tr>
<th>Event-Name</th>
<th>CHANNEL_BRIDGE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Core-UUID</td>
<td>17c1c070-8a13-11de-9ab6-91a5c9f91e77</td>
</tr>
<tr>
<td>FreeSWITCH-Hostname</td>
<td>jmesquita-dell</td>
</tr>
<tr>
<td>FreeSWITCH-IPv4</td>
<td>186.18.21.203</td>
</tr>
<tr>
<td>FreeSWITCH-IPv6</td>
<td>%3A%3A1</td>
</tr>
<tr>
<td>Event-Date-Local</td>
<td>2009-08-17%2003%3A20%3A17</td>
</tr>
<tr>
<td>Event-Date-GMT</td>
<td>Mon,%2017%20Aug%20009%2006%3A20%3A17%20GMT</td>
</tr>
<tr>
<td>Event-Date-Timestamp</td>
<td>1250490017110617</td>
</tr>
<tr>
<td>Event-Calling-File</td>
<td>switch_ivr_bridge.c</td>
</tr>
<tr>
<td>Event-Calling-Function</td>
<td>switch_ivr_multi_threaded_bridge</td>
</tr>
<tr>
<td>Event-Calling-Line-Number</td>
<td>847</td>
</tr>
<tr>
<td>Channel-State</td>
<td>CS_EXECUTE</td>
</tr>
<tr>
<td>Channel-State-Number</td>
<td>4</td>
</tr>
<tr>
<td>Channel-Name</td>
<td>sofla/internal/1001%40192.168.1.100</td>
</tr>
<tr>
<td>Unique-ID</td>
<td>07led3fa-8af6-11de-9ab6-91a5c9f91e77</td>
</tr>
<tr>
<td>Call-Direction</td>
<td>inbound</td>
</tr>
<tr>
<td>Presence-Call-Direction</td>
<td>inbound</td>
</tr>
<tr>
<td>Answer-State</td>
<td>answered</td>
</tr>
<tr>
<td>Channel-Read-Codec-Name</td>
<td>G722</td>
</tr>
<tr>
<td>Channel-Read-Codec-Rate</td>
<td>16000</td>
</tr>
<tr>
<td>Channel-Write-Codec-Name</td>
<td>G722</td>
</tr>
<tr>
<td>Channel-Write-Codec-Rate</td>
<td>16000</td>
</tr>
<tr>
<td>Caller-Username</td>
<td>1001</td>
</tr>
<tr>
<td>Caller-Dialplan</td>
<td>XML</td>
</tr>
<tr>
<td>Caller-Caller-ID-Name</td>
<td>1001</td>
</tr>
<tr>
<td>Caller-Caller-ID-Number</td>
<td>1001</td>
</tr>
<tr>
<td>Caller-Network-Addr</td>
<td>192.168.1.100</td>
</tr>
<tr>
<td>Caller-Destination-Number</td>
<td>1000</td>
</tr>
<tr>
<td>Caller-Unique-ID</td>
<td>07led3fa-8af6-11de-9ab6-91a5c9f91e77</td>
</tr>
<tr>
<td>Caller-Source</td>
<td>mod_sofia</td>
</tr>
<tr>
<td>Caller-Context</td>
<td>default</td>
</tr>
<tr>
<td>Caller-Channel-Name</td>
<td>sofla/internal/1001%40192.168.1.100</td>
</tr>
<tr>
<td>Caller-Profile-Index</td>
<td>1</td>
</tr>
<tr>
<td>Caller-Profile-Created-Time</td>
<td>1250490015373695</td>
</tr>
<tr>
<td>Caller-Channel-Created-Time</td>
<td>1250490015373695</td>
</tr>
<tr>
<td>Caller-Channel-Answered-Time</td>
<td>1250490017110617</td>
</tr>
<tr>
<td>Caller-Channel-Progress-Time</td>
<td>1250490015498382</td>
</tr>
<tr>
<td>Caller-Channel-Progress-Media-Time</td>
<td>1250490015453696</td>
</tr>
<tr>
<td>Caller-Channel-Hangup-Time</td>
<td>0</td>
</tr>
<tr>
<td>Caller-Channel-Transfer-Time</td>
<td>0</td>
</tr>
<tr>
<td>Caller-Screen-Bit</td>
<td>true</td>
</tr>
<tr>
<td>Caller-Privacy-Hide-Name</td>
<td>false</td>
</tr>
<tr>
<td>Caller-Privacy-Hide-Number</td>
<td>false</td>
</tr>
</tbody>
</table>
variable_write_codec: G722
variable_write_rate: 16000
variable_dialed_extension: 1000
variable_export_vars: dialed_extension
variable_ringback: %25(2000,4000,440.0,480.0)
variable_transfer_ringback: local_stream%3A/moh
variable_call_timeout: 30
variable_hangup_after_bridge: true
variable_continue_on_fail: true
variable_called_party_callgroup: techsupport
variable_current_application_data: user/1000%40192.168.1.100
variable_current_application: bridge
variable_dialed_user: 1000
variable_dialed_domain: 192.168.1.100
variable_local_media_ip: 186.18.21.203
variable_local_media_port: 31492
variable_switch_m_sdp: v=0
a=recvonly
a=sendonly
a=rtcp-fingerprint:0a0f
a=mid:bridge-a
a=mid:bridge-b
a=rtpmap:8000 PCMU/8000
a=rtpmap:8080 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtcp-fingerprint:0a0f

variable_endpoint_disposition: ANSWER
variable_signal_bond: 072a3ec0-8af6-11de-9ab6-91a5c9f91e77
variable_originate_disposition: SUCCESS

1.11 CHANNEL_UNBRIDGE event

A bridge has been terminated. The call itself will most probably be terminated since bridges exist during a call's lifespan.

TODO Check the example below for validity (for example, semicolons should be colons).

### CHANNEL_UNBRIDGE example

Channel-Read-Codec-Name; PCMU
Channel-Read-Codec-Rate; 8000
Channel-Write-Codec-Name; PCMU
Channel-Write-Codec-Rate; 8000
Caller-Username; jonas
Caller-Dialplan; XML
Caller-Caller-ID-Name; jonas
Caller-Caller-ID-Number; jonas
Caller-Network-Addr; 192.168.0.58
Caller-Destination-Number; 192.168.0.58
Caller-Unique-ID; 0dd4e4f7-36ed-a04d-a8f7-7aebb683af50
Caller-Source; mod_sofia
Caller-Context; default
Caller-Channel-Name; sofia/192.168.0.58/jonas@192.168.0.58:5060
Caller-Screen-Bit; yes
Caller-Privacy-Hide-Name; no
Caller-Privacy-Number; no
Originatee-Username; jonas
Originatee-Dialplan; XML
Originatee-Caller-ID-Name; jonas
Originatee-Caller-ID-Number; jonas
Originatee-Network-Addr; 192.168.0.58
Originatee-Destination-Number; 192.168.0.58/arne@192.168.0.58
Originatee-Unique-ID; f66e8e31-c9fb-9a0f-4e1-9a23a1586facb97f
Originatee-Source; mod_sofia
Originatee-Context; default
Originatee-Channel-Name; sofia/192.168.0.58/arne
Originatee-Screen-Bit; yes
Originatee-Privacy-Hide-Name; no
Originatee-Privacy-Number; no

1.12 CHANNEL_PROGRESS event

For outbound calls, the other party is in the alerting state; for inbound calls this party is alerting.
1.13 CHANNEL_PROGRESS_MEDIA event

See 1.12 CHANNEL_PROGRESS event section.

1.14 CHANNEL_OUTGOING event

An outgoing call is created.

1.15 CHANNEL_PARK event

A call is being parked in the PBX.

1.16 CHANNEL_UNPARK event

A call is being unparked.

1.17 CHANNEL_APPLICATION event

This event is generated by `mod_dptools:event`:

```
<action application='event' data='Event-Subclass=channel_state_change,State=checking_voicemail'/>
```

You can use this to trap some transitions happening in your calls.
CHANNEL_APPLICATION example

Content-Length: 1586
Content-Type: text/event-plain

Event-Subclass: channel_state_change
Event-Name: CHANNEL_APPLICATION
Core-UUID: d5cdce12-ec00-4ee8-97be-98bb734bb1fa
FreeSWITCH-Hostname: centos53_02005
FreeSWITCH-IPv4: 192.168.2.5
FreeSWITCH-IPv6: ::1
Event-Date-Local: 2010-02-23 00:15:37
Event-Date-GMT: Mon, 22 Feb 2010 15:15:37 GMT
Event-Date-Timestamp: 1266851737846113
Event-Calling-File: mod_dptools.c
Event-Calling-Function: event_function
Event-Calling-Line-Number: 981
State: checking_voicemail.
Channel-State: CS_EXECUTE
Channel-State-Number: 4
Channel-Name: sofia/internal/23702@domain23702.com
Unique-ID: 482c78ba-a2bf-4324-bf09-388b7b5fbb54
Call-Direction: inbound
Presence-Call-Direction: inbound
Answer-State: ringing
Channel-Read-Codec-Name: PCMU
Channel-Read-Codec-Rate: 8000
Channel-Write-Codec-Name: PCMU
Channel-Write-Codec-Rate: 8000
Caller-Username: 23702
Caller-Dialplan: XML
Caller-Caller-ID-Name: 23702
Caller-Caller-ID-Number: 23702
Caller-Network-Addr: 192.168.2.5
Caller-ANI: 23702
Caller-Destination-Number: 2000
Caller-Unique-ID: 482c78ba-a2bf-4324-bf09-388b7b5fbb54
Caller-Source: mod_sofia
Caller-Context: internal
Caller-Channel-Name: sofia/internal/23702@domain23702.com
Caller-Profile-Index: 1
Caller-Profile-Created-Time: 1266851737825925
Caller-Channel-Created-Time: 1266851737825925
Caller-Channel-Answered-Time: 0
Caller-Channel-Progress-Time: 0
Caller-Channel-Progress-Media-Time: 0
Caller-Channel-Hangup-Time: 0
Caller-Screen-Bit: true
Caller-Privacy-Hide-Name: false
Caller-Privacy-Hide-Number: false

1.18 CHANNEL_HOLD event

Triggers when a channel is put on hold either by using uuid_hold (see mod_commands) or receiving SDP with a=readonly.

CHANNEL_HOLD example

Event-Name: CHANNEL_HOLD
  .
  .
Channel-Call-State: HELD
  .
  .
1.19 CHANNEL_UNHOLD event
triggers after uuid_hold off <uuid> or receiving INVITE SDP with a=sendrecv

<table>
<thead>
<tr>
<th>CHANNEL_UNHOLD example</th>
</tr>
</thead>
<tbody>
<tr>
<td>Event-Name: CHANNEL_UNHOLD</td>
</tr>
<tr>
<td>.</td>
</tr>
<tr>
<td>Channel-Call-State: ACTIVE</td>
</tr>
<tr>
<td>.</td>
</tr>
</tbody>
</table>

1.20 CHANNEL_ORIGINATE event
Channel originate events are fired as soon as an originate (or bridge) completes.

1.21 CHANNEL_UUID event
This event indicates the Unique-ID of a channel has changed. The original ID will be reported by Old-Unique-ID.
This event will happen when you use the origination_uuid parameter of commands originate and bridge (see mod_commands).
2. System events

SHUTDOWN

Raised when FreeSWITCH started shutdown sequence.
MODULE_LOAD
Raised when module was load.

Content-Length: 480
Content-Type: text/event-plain

type: codec
name: LPC-10%202.4kbps
Event-Name: MODULE_LOAD
Core-UUID: 2130a7d1-c1f7-44cd-8fae-8ed5946f3aec
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2007-12-16%2022%3A24%3A56
Event-Date-GMT: Mon,%2017%20Dec%202007%2004%3A24%3A56%20GMT
Event-Date-timestamp: 1197865496783828
Event-Calling-File: switch_loadable_module.c
Event-Calling-Function: switch_loadable_module_process
Event-Calling-Line-Number: 174

MODULE_UNLOAD
Raised when module was unload.

type: application
name: lua
description: Launch%20LUA%20ivr
syntax: &lt;script&gt;
Event-Name: MODULE_UNLOAD
Core-UUID: ab0feafa-a9b0-4d77-b0a8-341d6b100b4f
FreeSWITCH-Hostname: vertux
FreeSWITCH-IPv4: 192.168.77.248
FreeSWITCH-IPv6: %::1
Event-Date-Local: 2008-12-11%2013%3A14%3A23
Event-Date-GMT: Thu,%2011%20Dec%202008%2012%3A14%3A23%20GMT
Event-Date-timestamp: 1228997663531389
Event-Calling-File: switch_loadable_module.c
Event-Calling-Function: switch_loadable_module_unprocess
Event-Calling-Line-Number: 524

RELOADXML
Raised when the xml configuration has been reloaded.

Event-Name: RELOADXML
Core-UUID: 6c6def18-9562-de11-a8e0-001fc6ab49e2
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: %::1
Event-Date-Local: 2009-06-26%2017%3A06%3A33
Event-Date-GMT: Fri,%2026%20Jun%202009%2021%3A06%3A33%20GMT
Event-Date-timestamp: 1246050393884782
Event-Calling-File: switch_xml.c
Event-Calling-Function: switch_xml_open_root
Event-Calling-Line-Number: 1917

NOTIFY
SEND_MESSAGE
RECV_MESSAGE
REQUEST_PARAMS
CHANNEL_DATA
GENERAL
COMMAND
SESSION_HEARTBEAT
RECV_EVENT
Event-Name: SESSION_HEARTBEAT
Core-UUID: 9b0de0b8-f55e-40d8-a2bd-179310b53493
FreeSWITCH-Hostname: myhost
variable_sip_user_agent: Zoiper rev.4688
variable_sip_via_host: 190.52.138.225
variable_sip_via_port: 60780
variable_sip_via_rport: 5060
variable_max_forwards: 70
variable_presence_id: 1000@192.168.0.2
variable_switch_r_sdp: v=0
o=Z 0 0 IN IP4 190.52.138.225
s=Z
c=IN IP4 190.52.138.225
t=0 0
m=audio 60790 RTP/AVP 3 110 98 8 0 101
a=rtpmap:3 GSM/8000
a=rtpmap:110 speex/8000
a=rtpmap:98 iLBC/8000
a=fmtp:98 mode=30
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

variable_remote_media_ip: 190.52.138.225
variable_write_codec: GSM
variable_write_rate: 8000
variable_local_media_ip: 192.168.0.2
variable_local_media_port: 19878
variable_endpoint_disposition: ANSWER
variable_current_application_data: $1-192.168.0.2@default
variable_current_application: conference
variable_conference_name: $1-192.168.0.2
variable_read_codec: L16
variable_read_rate: 8000

CLIENT_DISCONNECTED
SERVER_DISCONNECTED
SEND_INFO
RECV_INFO
CALL_SECURE
NAT
RECORD_START
RECORD_STOP
PLAYBACK_START
PLAYBACK_STOP

Received everytime a new playback starts. If multiple files has been set when calling the playback application, a PLAYBACK_START and
PLAYBACK_STOP event will be received for each file being played.

Also check out http://wiki.freeswitch.org/wiki/Misc.Dialplan_Tools_playback#Example_for_specific_playback_variables for information about how to set
specific variables that will follow the events.

PLAYBACK_STOP

See PLAYBACK_START.

CALL_UPDATE
Tells us to which UUID this channel was bridged to, through the "Bridged-To" header.
Event-Name: CALL_UPDATE

... 
Unique-ID: f3ebca6c-d9cd-4f89-a612-748e6e479dda
Bridged-To: ada7f3de-2374-4144-9b1d-eade29df0779
Direction: SEND
Channel-State: CS_EXCHANGE_MEDIA
Channel-Call-State: ACTIVE
Channel-Name: sofia/internal/sip:9998@192.168.56.1:56454
Call-Direction: outbound
Presence-Call-Direction: outbound
Channel-Presence-ID: 9998@192.168.56.2
Channel-Call-UUID: ada7f3de-2374-4144-9b1d-eade29df0779
Answer-State: answered
...

3. Other events

API

An API function has been invoked.

Api command documentation can be found in the mod_commands section.

Event-Name: API
Core-UUID: f3c23231-f251-49d8-bbf6-fe5c52af3762
FreeSWITCH-Hostname: fstest
FreeSWITCH-Switchname: fstest
FreeSWITCH-IPv4: 192.168.20.73
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2013-04-12%2012%3A01%3A31
Event-Date-GMT: Fri, %2012%20Apr%202013%2010%3A01%3A31%20GMT
Event-Date-Timestamp: 1365760891105542
Event-Calling-File: switch_loadable_module.c
Event-Calling-Function: switch_api_execute
Event-Calling-Line-Number: 2282
Event-Sequence: 2311
API-Command: version

BACKGROUND_JOB

Use this to receive an event when a job started with the bgapi call finishes. The BACKGROUND_JOB event will contain a Job-UUID that matches up with Job-UUID returned by the server when bgapi is called.

The following examples are in the context of an mod_event_socket client.

Listening for events

event plain BACKGROUND_JOB

Calling bgapi

bgapi originate sofia/mydomain.com/foo@bar.com &park()

Server response

Content-Type: command/reply
Reply-Text: +OK Job-UUID: e3054f48-151e-11dc-842a-d3a3942d3d63

Upon job completion, server response
It should be noted that the second Content-Length in the Background Job event indicates the length of the job uuid which is returned as a body within the event.

CUSTOM

See 2.2 Subclasses (or custom events) section.

RE_SCHEDULE

Reschedule a task in the PBX.

Event specific key/values:

Task-ID: 1
Task-Desc: heartbeat
Task-Group: core
Task-Runtime: 1178646608

HEARTBEAT

Status information for FreeSWITCH triggered by FreeSWITCH’s heartbeat every 20 seconds.

Event specific information:

DETECTED_TONE

Event sent when a tone detected.

Example when for event generated by `<action application="fax_detect"/>` when a fax tone detected:
Detected-Tone: fax
Event-Name: DETECTED_TONE
Core-UUID: 5859d2de-ccce-11dc-aab0-69b2875ec123
FreeSWITCH-Hostname: abacus
FreeSWITCH-IPv4: <myip>
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2008-01-27%2017%3A19%3A30
Event-Date-GMT: Sun,27%20Jan%202008%2016%3A19%3A30%20GMT
Event-Date-Timestamp: 1201450770979522
Event-Calling-File: switch_ivr_async.c
Event-Calling-Function: tone_detect_callback
Event-Calling-Line-Number: 1098

ALL

Will show all events, including custom events. There is no such event like ALL. This is just like macro when specifying which events to receive.

6. Undocumented events

LOG

INBOUND_CHAN

OUTBOUND_CHAN

STARTUP

PUBLISH

UNPUBLISH

TALK

Triggered when speech is detected on channel. Needs parameter "vad" to be set in the sip profile. e.g:

<param name="vad" value="both"/>

You also need to set some channel variables to make this work:

<action application="export" data="fire_talk_events=true"/>
<action application="export" data="fire_not_talk_events=true"/>

Event-Name: TALK
Core-UUID: da0f9ecb-5e56-4be6-891d-a6c7b86c98f6
FreeSWITCH-Hostname: fstest
FreeSWITCH-Switchname: fstest
FreeSWITCH-IPv4: 192.168.20.73
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2013-04-12%2012%3A10%3A36
Event-Date-GMT: Fri,12%20Apr%202013%2010%3A10%3A36%20GMT
Event-Date-Timestamp: 1365761436805557
Event-Calling-File: switch_rtp.c
Event-Calling-Function: rtp_common_write
Event-Calling-Line-Number: 5329
Event-Sequence: 532

NOTALK

Triggered when speech is off for the channel.

Event-Name: NOTALK
Core-UUID: da0f9ecb-5e56-4be6-891d-a6c7b86c98f6
FreeSWITCH-Hostname: fstest
FreeSWITCH-Switchname: fstest
FreeSWITCH-IPv4: 192.168.20.73
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2013-04-12%2012%3A10%3A36
Event-Date-GMT: Fri,12%20Apr%202013%2010%3A10%3A36%20GMT
Event-Date-Timestamp: 1365761436805557
Event-Calling-File: switch_rtp.c
Event-Calling-Function: rtp_common_write
Event-Calling-Line-Number: 5329
Event-Sequence: 532

SESSION_CRASH
Channel-State: CS_EXECUTE
Channel-State-Number: 4
Channel-Name: sofia/default/1006@10.0.1.250:5060
Unique-ID: 8dcbb29e-b349-462a-84ca-b0ec73681284
Call-Direction: inbound
Answer-State: answered
Channel-Read-Codec-Name: G722
Channel-Read-Codec-Rate: 16000
Channel-Write-Codec-Name: G722
Channel-Write-Codec-Rate: 16000
Caller-Username: 1006
Caller-Dialplan: XML
Caller-Caller-ID-Name: Brian%20West
Caller-Caller-ID-Number: 1006
Caller-Network-Addr: 10.0.1.240
Caller-Destination-Number: 9999
Caller-Unique-ID: 8dcbb29e-b349-462a-84ca-b0ec73681284
Caller-Source: mod_sofia
Caller-Context: default
Caller-Channel-Name: sofia/default/1006@10.0.1.250:5060
Caller-Channel-Created-Time: 1197864491030187
Caller-Channel-Answered-Time: 1197864491080700
Caller-Channel-Hangup-Time: 0
Caller-Channel-Transfer-Time: 0
Caller-Screen-Bit: yes
Caller-Privacy-Hide-Name: no
Caller-Privacy-Hide-Number: no
variable_sip_authorized: true
variable_sip_mailbox: 1006
variable_sip_auth_username: 1006
variable_sip_auth_realm: 10.0.1.250
variable_mailbox: 1006
variable_accountcode: 1006
variable_presence_id: 1006%4010.0.1.250
variable_user_context: default
variable_effective_caller_id_name: Extension%201006
variable_effective_caller_id_number: 1006
variable_sip_from_user: 1006
variable_sip_from_port: 5060
variable_sip_from_uri: 1006%4010.0.1.250%3A5060
variable_sip_from_host: 10.0.1.240
variable_sofia_profile_name: default
variable_sip_req_user: 9999
variable_sip_req_uri: 9999%4010.0.1.250%3A5060
variable_sip_req_to_user: 9999
variable_sip_req_to_port: 5060
variable_sip_req_to_uri: 9999%4010.0.1.250%3A5060
variable_sip_request_time: 10.0.1.250
variable_sip_contact_user: 1006
variable_sip_contact_port: 5060
variable_sip_contact_uri: 1006%4010.0.1.250%3A5060
variable_sip_contact_host: 10.0.1.250
variable_sip_contact_transport: tcp
variable_sip_contact_transport_host: 10.0.1.250
variable_sip_contact_transport_port: 5060
variable_sip_contact_transport_user: 1006
variable_sip_via_host: 10.0.1.250
variable_sip_via_port: 5060
variable_switch_r_sdp: v=0
a=sendrecv
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
variable_remote_media_ip: 10.0.1.240
variable_remote_media_port: 2230
variable_read_codec: G722
variable_read_rate: 16000
variable_write_codec: G722
variable_write_rate: 16000
variable_use_profile: nat
variable_numbering_plan: US
variable_default_gateway: asterlink.com
variable_default_area_code: 918
variable_user_name: default
variable_domain_name: default
variable_local_media_ip: 10.0.1.250
variable_local_media_port: 2230
variable_endpoint_disposition: ANSWER
DTMF-Digit: 1
DTMF-Duration: 2000
Event-Name: DTMF
Core-UUID: 2130a7d1-11f7-44cd-8fae-8ed5946f3cec
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2007-12-16T22:27:42
Event-Date-GMT: Mon, 17 Dec 2007 04:27:42 GMT
Event-Date-timestamp: 1197865662745906
Event-Calling-File: switch_channel.c
Event-Calling-Function: switch_channel_dequeue_dtmf
Event-Calling-Line-Number: 269

MESSAGE
Contains the composed text of the message much like an email.

Content-Length: 699
Content-Type: text/event-plain

sip_mailbox: 1006
domain_name: 10.0.1.250
from: 1006@10.0.1.250
proto: sip
rpid: unknown
status: Registered
type: MESSAGE

PRESENCE_IN
Content-Length: 559
Content-Type: text/event-plain

proto: sip
login: sip%3Amod_sofia%4010.0.1.250%3A5060
rpid: unknown
from: 1006@10.0.1.250
status: Registered
event_type: PRESENCE_IN

PRESENCE_OUT

PRESENCE_PROBE
proto: sip
login: sip@mod_sofia@10.0.1.250@5060
from: 1009@10.0.1.250
status: Available
event_type: presence
event_subtype: probe
proto-specific-event-name: dialog
Event-Name: PRESENCE_PROBE
Core-UUID: 2130a7d1-clf7-44cd-8fae-8ed5946f3cec
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2007-12-16%2022%3A31%3A16
Event-Date-GMT: Mon,%2017%20Dec%202007%2004%3A31%3A16%20GMT
Event-Date-timestamp: 1197865876565022
Event-Calling-File: sofia_presence.c
Event-Calling-Function: sofia_presence_sub_reg_callback
Event-Calling-Line-Number: 484

MESSAGE_WAITING
Contains the protocol specific MWI data.
	sendevent MESSAGE_WAITING
MWI-Messages-Waiting: yes
MWI-Message-Account: jonas@gauffin.com
MWI-Voice-Message: 2/1 (1/1)
Voice messages: total_new_messages / total_saved_messages (total_new_urgent_messages / total_saved_urgent_messages)

MESSAGE_QUERY
Content-Length: 470
Content-Type: text/event-plain
Message-Account: sip@mod_sofia@10.0.1.250
Event-Name: MESSAGE_QUERY
Core-UUID: 2130a7d1-clf7-44cd-8fae-8ed5946f3cec
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2007-12-16%2022%3A29%3A59
Event-Date-GMT: Mon,%2017%20Dec%202007%2004%3A29%3A59%20GMT
Event-Date-timestamp: 1197865799573052
Event-Calling-File: sofia_reg.c
Event-Calling-Function: sofia_reg_handle_register
Event-Calling-Line-Number: 603

ROSTER
Content-Length: 457
Content-Type: text/event-plain
proto: sip
from: 1006@10.0.1.250
Event-Name: ROSTER
Core-UUID: 2130a7d1-clf7-44cd-8fae-8ed5946f3cec
FreeSWITCH-Hostname: localhost.localdomain
FreeSWITCH-IPv4: 10.0.1.250
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2007-12-16%2022%3A32%3A29
Event-Date-GMT: Mon,%2017%20Dec%202007%2004%3A32%3A29%20GMT
Event-Date-timestamp: 1197865949889095
Event-Calling-File: sofia_reg.c
Event-Calling-Function: sofia_reg_handle_register
Event-Calling-Line-Number: 583

RECV_RTCP_MESSAGE
CODEC

DETECTED_SPEECH

PRIVATE_COMMAND

TRAP

generic event that can be used to indicate a severe error

ADD_SCHEDULE

generated when using sched__api command
Event-Name: ADD_SCHEDULE
Core-UUID: f3c23231-f251-49d8-bbf6-fe5c52af3762
FreeSWITCH-Hostname: fstest
FreeSWITCH-Switchname: fstest
FreeSWITCH-IPv4: 192.168.20.73
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2013-04-12%2011%3A56%3A24
Event-Date-GMT: Fri,%2012%20Apr%202013%2009%3A56%3A24%20GMT
Event-Date-Timestamp: 1365760584365548
Event-Calling-File: switch_scheduler.c
Event-Calling-Function: switch_scheduler_add_task
Event-Calling-Line-Number: 222
Event-Sequence: 2252
Task-ID: 4
Task-Desc: sched_api_function
Task-Group: none
Task-Runtime: 1365760589

DELE_SCHEDULE

generated when using sched_del or an scheduled task is finished

Event-Name: DEL_SCHEDULE
Core-UUID: f3c23231-f251-49d8-bbf6-fe5c52af3762
FreeSWITCH-Hostname: fstest
FreeSWITCH-Switchname: fstest
FreeSWITCH-IPv4: 192.168.20.73
FreeSWITCH-IPv6: %3A%3A1
Event-Date-Local: 2013-04-12%2011%3A56%3A24
Event-Date-GMT: Fri,%2012%20Apr%202013%2009%3A56%3A24%20GMT
Event-Date-Timestamp: 1365760589045539
Event-Calling-File: switch_scheduler.c
Event-Calling-Function: switch_scheduler_execute
Event-Calling-Line-Number: 74
Event-Sequence: 2254
Task-ID: 4
Task-Desc: sched_api_function
Task-Group: none
Task-Runtime: 1365760589

EXE_SCHEDULE