Asterisk SIP Settings

User Guide

Schmooze Com Inc.

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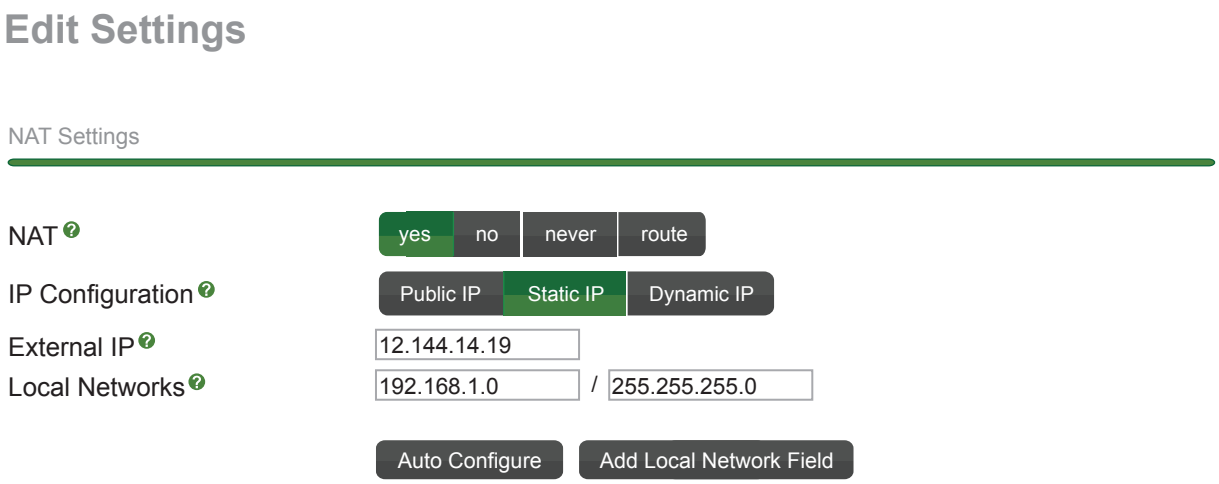
Overview

It was necessary in the past to hand edit files like "/etc/asterisk/sip_nat.conf" as part of the initial installation of any Asterisk based deployment. With the introduction of the Asterisk SIP Settings module, most SIP settings are made available in the GUI. This includes the all important NAT, External IP, Local Network, Enabled Codecs and Codec order.

Logging In

- Log into the Asterisk SIP Settings module and you should see a screen like this.



NAT Settings

- NAT settings are crucial in avoiding issues with one-way audio. You may need to adjust these settings to match the network environment in which you place your system.
    - **NAT**- (Network Address Translation) The method used to pass network traffic between the public and private IP networks. We recommend setting this to "Yes" if you are going to have SIP trunks or remote phones on a different network then the PBX.
        - **Yes Mode**- This will always ignore info and assume NAT.
        - **No Mode**- This will use NAT only according to RFC3581.
        - **Never Mode**- This will never attempt to use NAT mode or RFC3581.
        - **Route Mode**- This will assume NAT and not send rport.
    - **IP Configuration**- Indicate whether the system has a public IP or requires NAT setting. Automatic configuration of what is often set as "sip_nat.conf."
    - **External IP**
        - **(Set as static IP - asterisk:externip)**- External static IP or FQDN as seen on the WAN side of the router.

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- **(Set as dynamic IP - asterisk:externhost)**- External FQDN as seen on the WAN side of the router and updated dynamically, i.e. mydomain.dyndns.com.

◦ **Local Networks**

- **(asterisk:localnet)**- Local network settings in the form of "ip/mask" such as, "192.168.1.0/255.255.255.0." For networks with more than one LAN subnet, such as VPN network, us the "Add Local Network" button to add more fields. Blank fields will be removed upon submitting.

Audio Codecs

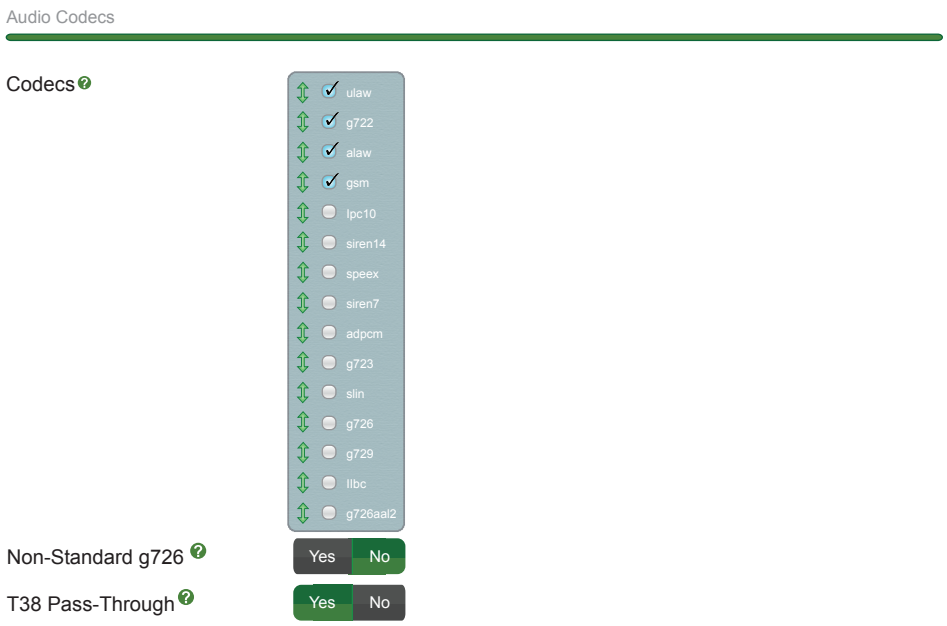
- Check the desired codecs and drag to reorder. All others will be disabled unless explicitly enabled in a device or trunk configuration. Note that some codecs, such as g729, require commercial licensing.

◦ **Non Standard g726**

- **(asterisk:g726nonstandard)**- If the peer negotiates G726-32 audio, use AAL2 packing order instead of the RFC3551 packing order. This is required for Sipura and GRandstream ATAs, among others. Contrary to the RFC3551 specification, the peer should be negotiating AAL2-G726-32 instead.

◦ **T38 Pass Through**

- **(asterisk:t38pt_udptl)**- Enables T38 pass-through. This is used with SIP channels that support sending and receiving of T38 FAX codecs to pass the call. Asterisk cannot process the media.



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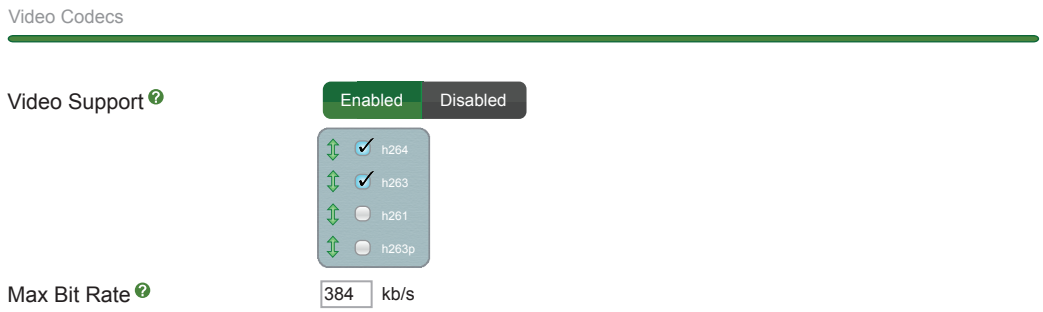
Video Codecs

- Video support can be enabled or disabled (default is disabled). When enabled, the following settings are available.
  - **Codecs**- As with audio codecs, check the desired codecs and drag to reorder. All others will be disabled.
  - **Max Bit Rate**- Maximum bit rate for video calls in kb/s.



Media & RTP Settings

- Change the way media is handled during calls.
  - **Reinvite Behavior**
    - **(asterisk:canreinvite)**
      - **Yes**- This uses standard reinvites.
      - **No**- This equals never issue reinvites.
      - **Nonat**- This is an additional option to allow media path redirection (reinvite), but only when the peer that the media is being sent to is not behind NAT. The RTP core can determine it based on the apparent IP address the media arrives from.
      - **Update**- Use UPDATE for media path redirection instead of INVITE (yes = update + nonat).
  - **RTP Timers**
    - **(asterisk:rtptimeout)**- Terminate call if "rtptimeout in seconds" of no RTP or RTCP activity on the audio channel when the call is not on hold. This enables the ability to hang-up a call in the case of the phone disappearing from the net, like a power outage or someone tripping over a cable. Default is 30 sec.
    - **(asterisk:rtpholdtimeout)**- Terminate call if "rtpholdtimeout in seconds" of no RTP or RTCP activity on the audio channel when the call is on hold. Note that this value must be greater then the "rtptimeout" value. Default is 300 sec.
    - **(asterisk:rtpkeepalive)**- Send "keepalives" in the RTP stream to keep NAT open during periods when no RTP stream may be flowing (like on hold). Default is 0.

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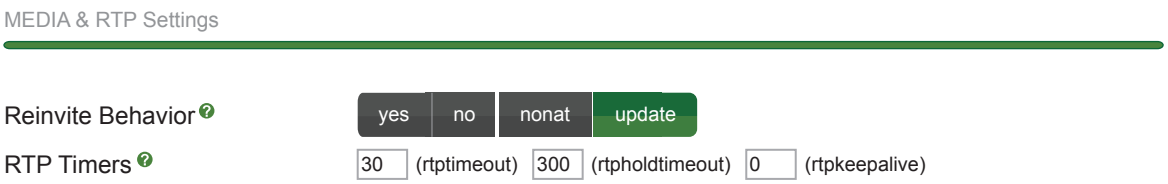
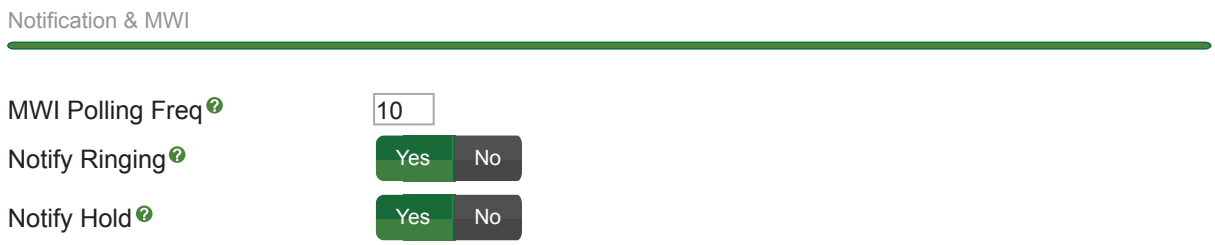


Notification & MWI

- Used to control polling frequency for MWI (Message Waiting Indication) and in controlling INUSE information.
  - **MWI Polling Freq**- Interval in seconds to check if the MWI state has changed and inform peers. Default is 10 sec.
  - **Notify Ringing**- Control whether the subscription already INUSE get sent ringing when another call is sent. Useful when using BLF. Default is "Yes."
  - **Notify Hold**- Control whether the subscription INUSE gets sent ONHOLD when the call is placed on hold. Useful when using BLF. Default is "Yes."



Registration Settings

- Control registration settings. Note that the defaults are generally sufficient.
  - **Registrations**
    - **(asterisk:registertimeout)**- Retry registration attempts every XX seconds until successful or XX registration attempts have been made. Default is 20 sec.
    - **(asterisk:registerattempts)**- Number of times to try and register before giving up. A value of 0 means to keep trying forever. Normally you would leave this as the default of 0 so that Asterisk will continue trying to register until successful in the case of network or gateway outages.
  - **Registration Times**
    - **(asterisk:minexpiry)**- Minimum length of registrations/subscriptions. Default is 60 sec.
    - **(asterisk:maxexpiry)**- Maximum allowed time of registrations. Default is 3600 sec.
    - **(asterisk:defaultexpiry)**- Default length of incoming and outgoing registrations. Default is 120 sec.

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Registration Settings

Registrations	20	(registertimeout)	0	(registerattempts)		
Registration Times	60	(minexpiry)	3600	(maxexpiry)	120	(defaultexpiry)

Jitter Buffer Settings

- Jitter buffer temporarily stores arriving packets in order to minimize delay.
  - **Jitter Buffer**
    - **(asterisk: jbenable)**- Enable or disable the use of a jitter buffer on the receiving side of a SIP channel. An enabled jiter buffer will only be used if the sending side can create a jitter buffer and the receiving side cannot accept jitter. The SIP channel can accept jitter, thus a jitter buffer on the receiving SIP side will be used only if it is forced and enabled. For example, if receiving from a jittery channel to voicemail, the jitter buffer will be used if enabled. However, it will not be used when sending to a SIP endpoint since they usually have their own jitter buffers. See “jbforce” to force it’s use always.
  - **Force Jitter Buffer**
    - **(asterisk:jbforce)**- Forces the use of a jitter buffer on the receiving side of a SIP channel. Normally the jitter buffer will not be used if receiving a jittery channel, rather the jitter will be passed on to another SIP channel such as an endpoint that typically has a jitter buffer. This will force the use of the jitter buffer before sending the stream on. This is generally not recommended as it adds additional latency into the stream.
  - **Implementation**
    - **(asterisk:jbimpl)**- Jitter buffer implementation used on the receiving side of a SIP channel. Two implementations are available:
      - **Fixed**- This always equals “jbmaxsize.”
      - **Adaptive**- This uses variable size (the new jb of IAX2).
  - **Jitter Buffer Logging**
    - **(asterisk:jblog)**- Enable or disable jitter buffer frame logging.
  - **Jitter Buffer Size**
    - **(asterisk:jbmaxsize)**- Max length of the jitter buffer in milliseconds. Default is 200.
    - **(asterisk:jbresyncthreshold)**- Jump in the frame timestamps over which the jitter buffer is resynchronized. Useful to improve the quality of voice with bug jumps in broken timestamps. This anomaly is usually sent from exotic devices and programs.

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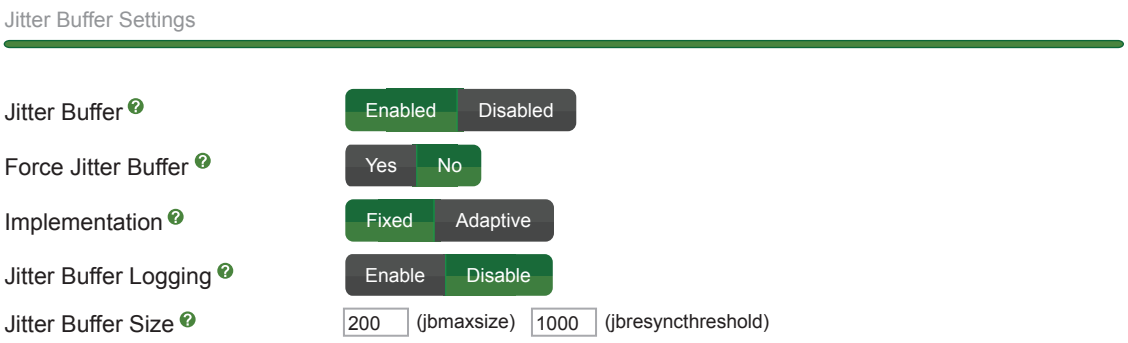
Default is 1000 and it can be set to -1 to disable.



Advanced General Settings

- Change the way media is handled during calls.
  - **Language**
    - **(asterisk:language)**- The default language for a channel.
  - **Default Context**
    - **(asterisk:context)**- Default context for incoming calls if not specified. PBXact sets this to "from-sip-external," which is used in conjunction with anonymous SIP calls. If you change this, it will effect that behavior. It is recommended that you leave this blank.
  - **Bind Address**
    - **(asterisk:bindaddr)**- The IP address to bind to and listen for calls on the bind port. If set to 0.0.0.0, Asterisk will listen on all addresses. We recommend you leave this blank. Asterisk 1.8 and above supports IPv6 and an address of "::" will listen on both IPv4/IPv6.
  - **Bind Port**
    - **(asterisk:bindport)**- This is the local incoming UDP port that Asterisk will bind to and listen for SIP messages. The SIP standard is 5060 and in most cases you will use this default. It is recommended that you leave this blank.
  - **Allow SIP Guests**
    - **(asterisk:allowguest)**- When set, Asterisk will allow guest SIP calls and send them to the default SIP contact. Turning this off will keep anonymous SIP calls from entering the system. However, the "Allow Anonymous SIP Calls" from the General Settings section will not function. Allowing guest calls but rejecting the "Anonymous SIP Calls" in the General Settings section will enable you to see the call attempts and debug incoming calls that may be misconfigured and appearing as guests.

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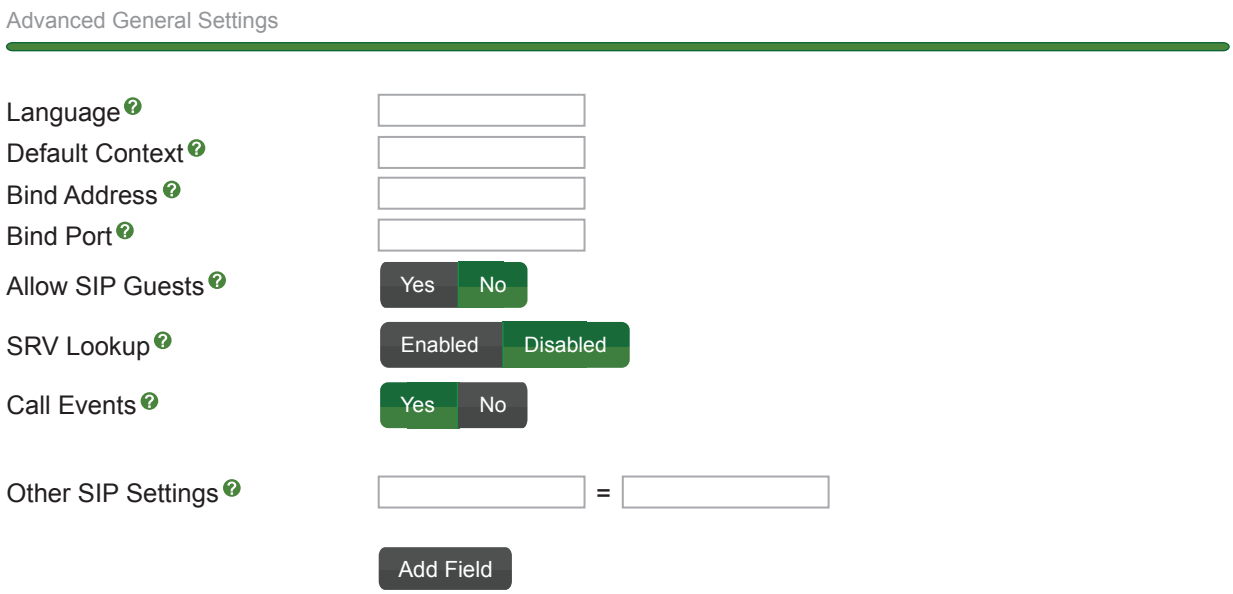
- **SRV Lookup**
  - **(asterisk:srvlookup)**- Enable or disable. See current version of Asterisk for limitations on SRV functionality.
- **Call Events**- Generate manager events when SIP performs events such as hold.
- **Other SIP Settings**- You may set any other SIP settings not present here that are allowed to be configured in the General Settings section of "sip.conf." There will be no error checking against these settings, so check them carefully. They should be entered as "[setting]=[value]." Click the "Add Field" button for additional lines. Blank lines will be deleted when you press "Submit."



- Don't forget to press the "Submit Changes" button when done.

Recap

- The Asterisk SIP Settings module affords easy access to the all-important SIP settings such NAT, External IP, Local Network, Enabled Codecs and Codec order. It is important to understand the ramifications of the various SIP settings as they greatly affect voice operations and quality.

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