Sangoma Technologies Corp.

Netborder Express Gateway

Release Notes

Release 4.4.3

November 22, 2013

Netborder Express Gateway Release Notes

1 Product Compatibility

Here are some of the major compatibility points.

- Operating Systems Supported:
  - Linux CentOS 5.0 to 6.2 32 bit
  - Linux Ubuntu 10.04 32 bit
- Sangoma Telephony Cards Supported:
  - AFT A101/2/4/8 T1/E1/J1 with hardware echo cancellation (PCI/PCI-Express)
  - AFT A200 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
  - AFT A400 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
  - AFT A500 ISDN-BRI with hardware echo cancellation (PCI/PCI-Express)
  - AFT B600 Analog FXO/FXS with hardware echo cancellation (PCI/PCI-Express)
  - AFT B700 Combo ISDN-BRI/Analog FXS with hardware echo cancellation (PCI/ PCI-Express).
- Sangoma Software Release Versions Supported:
  - Driver : 7.0.2.0 (included in gateway software package)
  - Firmware
    - A101D : v37
    - A102D : v37
    - A104D : v43
    - A108D : v43
    - A116D : v44
    - A200D : v13
    - A400D: v11
    - A500 : v35
    - B500 : v43
    - B600 : v03
    - B700 : v35
- SIP 3261 compliant endpoints using either TCP, TLS or UDP as the transport protocol
- DTMF relay as per IETF RFC 2833.
- RTP/RTCP as per IETF RFC 3550/3551
- SRTP as per IETF RFC 3711
- Minimum Server requirements:
  - Less than 2 T1/E1s : Dual core processor (minimum 4MB cache) or better with a minimum 1 GB of RAM.
  - Up to 8 E1/T1: Quad core processor (minimum 6MB cache) or better with a minimum 2 GB of RAM.
  - Up to 16 E1/T1: Dual Quad core processor (minimum 8MB cache) or better with a minimum of 2GB of RAM

Feature Support

Here is a brief list of the product features as supported in this release

Feature	Notes
PSTN-initiated calling	Support FXO & FXS analog interface Support ISDN-PRI Q931 (DMS100, 4ESS, 5ESS, National ISDN 2, ISDN-Japan, EuroISDN) terminal and network sides NFAS for DMS100, 4ESS, 5ESS and National ISDN 2 variants terminal and network sides NFAS with D-Channel backup is not supported Support ISDN-BRI terminal and network sides (point-to-point and multipoint) Advice of charge is not supported ISDN Overlap Dialing Reception supported ISDN-R2 signaling (Central and Southern America) terminal and network sides ISDN-CAS signaling is not supported
SIP-initiated calling	The Gateway listens on port 5066 by default
Support for 3xx redirect primitives	Includes “hybrid” redirect (redirecting to either SIP or PSTN endpoint)
SIP Registration	Allows to register the gateway to a third party SIP registrar
SIP HTTP Digest Authentication as per RFC2617 and RFC3261 section 22.4	Authenticate on INVITE and REGISTER requests only
TLS v1.0 as per RFC3261 SIP/TLS requirements	Authenticate and establish a secure link with SIP endpoints
RTP processing as per RFC 3550 and RTCP as per RFC 3551	G.711 codecs (uLaw and A-law) with law conversion
Secure RTP (SRTP) as per RFC 3711 and	Exchange secure audio between endpoints
DTMF per RFC 2833	Both DTMF relay (PSTN to SIP) and DTMF re-generation (SIP to PSTN)
Fax relay as per ITU-T.38	Allows sending FAX across VoIP networks UDPTL transport only ITU-T V.34 is not supported
Mapping of PSTN calls to SIP endpoints through rules, including DNIS-based routing	Configurable routing rules

Mapping of SIP calls to PSTN ports, trunks and DN through rules	Configurable routing rules
CallerID/ANI/DNIS and custom information element	Available in SIP message
Packaged as Linux Service	
FXO interface calibration	Allows to calibrate the FXO interfaces on order to minimize the line echo
Integrated with syslog	
Configurable logging per sub-system	
Call logs	Per call information
Web Service Interface for management	
Web User Interfaces for configuration and management	Allows to monitor and configure the gateway Contextual help
SNMP	DS0 & DS1 Mibs (RFC 2494 & 1232) and proprietary gateway call statistics MIB

2 Acquiring a License

The Gateway is licensed on a per telephony port basis. The license is host locked.

To obtain a **full multi-port license**, simply obtain the *MAC (Media Access Control)* address of the server and contact our support organization at 1800 388 2475 or direct at +1 905 474 1990 or email at techdesk@sangoma.com.

To get the physical address of the Ethernet adapter, simply start a shell command prompt and execute the following command: “ifconfig -a”. Then look for the Physical Address item. It would look something like: 00-0B-DB-D8-06-00.

3 Changes Since Last Release

3.1 Release 4.4.3 Patch Release

- **Fixed bugs**

  Bug R9471: Overlap Dialing: Expose T302 timer value configuration in PSTN config.

3.2 Release 4.4.2 Patch Release

- **Fixed bugs**

  Bug R7925: Overlap Dialing Support for PR

  Added support for IE11 web browsers

3.3 Release 4.4.1 Patch Release

3.4 Release 4.4.1 Patch Release

- **Fixed bugs**

  Bug R8999: Channel Not Releasing

3.5 Release 4.4.0 Maintenance Release

- **New Features**

  Validated Lync 2013 Certification Compatibility.

3.6 Release 4.3.15 Maintenance Release

- **New Features**

  Integrated Drivers version 7.0.2.0.

  Added Support for digital A116 boards.

3.7 Release 4.3.14 Maintenance Release

- **New Features**

  Added support of overlap dialing reception for PRI channels.

3.8 Release 4.3.13 Maintenance Release

- **New Features**

  Added Support of DTMF CallerID detection for analog FXO channels.

3.9 Release 4.3.12 Maintenance Release

- **New Features**

  Integrated linux driver version 3.5.25.4

3.10 Release 4.3.11 Maintenance Release

- **Fixed bugs**

  Bug R6671: Problem capturing information from Referred-By: header in INVITE from Lync

3.11 Release 4.3.10 Maintenance Release

- **Fixed bugs**

  Bug R7304: Patch to release call waiting SETUP messages for BRI channels

3.12 Release 4.3.9 Maintenance Release

- **Fixed bugs**

  Bug R6442: sip.in.referredBy* parameters are not extracted for INVITE requests

3.13 Release 4.3.8 Maintenance Release

- **Fixed bugs**

  Bug R6405: WARN - Unexpected eBOARD_CALL_ACCEPTED in state CALLENG_ACCEPTED

3.14 Release 4.3.7 Maintenance Release

- **Fixed bugs**

  Bug R6194: NBE Call Rejection Resources Unavailable.

3.15 Release 4.3.6 Maintenance Release

- **Fixed bugs**

  Bug R6230: Updated Norway Tones definition.

3.16 Release 4.3.5 Maintenance Release

- **New Features**
  - Added PSTN channel name to CDR logs.
- **Fixed bugs**
  - Bug R5104: Mexico CAS R2 MF is not working in Mexico

3.17 Release 4.3.4 Maintenance Release

- **Fixed bugs**
  - Bug R5021: Analog Call-Control slowing down real-time media threads under load.
  - Bug 2539: Got ERROR SangomaAnalogBChannel::writeBChannelsDataSangoma::write() failed
  - Bug R5014: Netborder light debian package uninstalls Wanpipe driver when updating

3.18 Release 4.3.3 Maintenance Release

- **Fixed bugs**

- Bug R4723: NBE system failure due to a race condition under heavy load while logging SIP messages.
- Bug R4817: ISDN Inband PROGRESS tone indicator selector in the WEB UI does not display the BOTH choice when configured in 4ESS or 5ESS.

3.19 Release 4.3.2 Maintenance Release

- **Fixed bugs**
  - Bug R3813: wizard.properties file does not contains the proper variable to refer to installation directory.
  - Bug R3793: Add configuration parameter to A108 card to control port numbering to match Sangoma DS3 mux numbering.

3.20 Release 4.3.1 Maintenance Release

- **Fixed bugs**
  - Bug R3483: Quick Setup Wizard - tab key not working correctly when FXO ports are installed.

3.21 Release 4.3.0 Maintenance Release

- **New features**
  - Support of ISDN-BRI Overlap Dialing reception for PSTN inbound calls.
  - BRI span is now moved into an alarmed state when an outbound call on a channel, belonging to that span, fails because of a failure to bring layer 1 or 2 up (considered as cable unplugged).
  - Support package generation tool in the Web UI.
  - Support for UK caller Id on FXO before 1st without polarity reversal.
- **Fixed bugs**
  - Bug R3389 fixed : It is now possible to detect FSK caller-id even if there is not at least 500 ms of silence before the FSK data.
  - Bug R3428: NBE BRI gateway does not ACK ISDN layer 3 RESTART messages.
  - Bug R3382: Generate a ring back tone when we get a 183.
  - Bug R3212: Improve BRI interface state management.

3.22 Release 4.2.3 Maintenance Release

- **New features**
  - Debian package is now available for Debian/Ubuntu.
  - New Linux Wanpipe driver 3.5.25.4
- **Fixed bugs**
  - Bug R2952 fixed : It is now possible to import configuration on Ubuntu.
  - Bug R3222 fixed : It is now possible to import configuration
  - Bug R2870 fixed : install-cron-job.sh do not generate error anymore when starting netborder-gateway service on Ubuntu.

- Bug R3363 fixed : BRI interface may some time fails to mount the physical layer

3.23 Release 4.2.2 Maintenance Release

- **Fixed bugs**
  - Bug R2972 – A200 FXO fails to dial out.

3.24 Release 4.2.1 Maintenance Release

- **New features**
  - Support of B500 BRI boards with hardware echo-canceller.
  - Support of A400 FXS/FXO boards with hardware echo-canceller.
  - Analog line status are now shown one line per board in order to save display space.
  - Character “,” supported in phone numbers to pause dialling for 2 seconds (analog FXO - ref. 6942).
- **Fixed bugs**
  - Bug 6646 - IE fails to parse audio_form.js when opening Quick Setup tab.
  - Bug 6940 - Quick Setup lightbox does not work.
  - Bug 6947 - ISDN Cause Information Element is not decoded properly when the recommendation field is present.
  - Bug 6905 - SNMP gwCallDetailTable clears after every 15 minutes.
  - Bug 6958 - SIP port defaults to 5060 in TLS (instead of expected 5061).
  - Bug 6613 - Decreasing “Failed to obtain call reference” message to DEBUG level as it only affects trace logging.
  - Bug 6500 - R2 no local ringback generated towards PSTN on inbound calls - ringback gen and early media tested.
  - Bug 6611 - Indicate in online help that BRI Point-To-Multi point is not supported in NETWORK mode.
  - Bug 6526 - Report correct SNMP data for CAS/R2 calls + gwNumActivePstnInCalls and OutCalls SNMP call stats were inverted for all PSTN protocols.
  - Bug 6598 - Import/Export and Provisioning configuration is out of date.
  - Bug 6514 - FXO Calibration start page text clipping in lightbox.
  - Bug 6462 - NBE should send CANCEL after SIP INVITE Timeout.
  - Bug 6516 - DTMF digit duration has the wrong help text.
  - Bug 6563 - D-Channel pcap trace not flushed on Linux.
  - Bug 6097 - Add lightbox page for Advanced configuration and Logging tabs.
  - Bug 6645 - Add support for Firefox 5+.
  - Bug 6580 - GW Dropping Call Park Music on Hold Media.
  - Bug 6623 - IE 9.0 freezes in restart after confirming gateway restart.
  - Bug 6569 - Missing endline in log text.
  - Bug 6654 - Importing configuration of the same version when version has a patch fails.

- ○ Bug 6511 - Export fail because of an IndexError exception.
- ○ Bug 6660 - lightbox_logging and lightbox_provisionning are broken.
- ○ Bug 6667 - Missing kernel-devel requirements in linux rpm.
- ○ Bug R2181 - Don't disconnect call upon reception of progress with cause 127
- ○ More robust detection of disconnect tones on long-lasting calls.

3.25 Release 4.2.0 Limited Availability Release

- New Features
  - Modified gateway manager configuration wizard to remember entered configuration parameters used to generate configuration files. When the wizard is run again, the wizard proposed the rememberred values in the wizard screens. The pstn-config.xml and routing-rules.xml files be overwritten according to the values specified while going through the wizard steps.
  - Now provide a set of 2 rpm files that could be used to install the Netborder Express Gateway:
    - netborder-express-gateway.rpm : Default RPM that should be used for most installations
    - netborder-express-gateway-ligth.rpm: RPM that couldl be used by OEM vendors who deploy their application on one hardware platform. This RPM does not required C compilier and kernel sources, However, it requires a wanpipe RPM package specially generated for the hardware platform used by the OEM vendor. Consult the $INSTALLDIR/doc/Driver_RPM_Generation_Guide.pdf for more details.
- Configuration parameters changes
  - Configuration wizard parameters are remembered in a new file $INSTALLDIR/config/wizard.properties.
  - Added the following parameters in oamWebServer.proprerties:
    - netborder.oamWebServer.wizardPropertiesFile
    - netborder.oamWebServer.wizardParamDB
  - Moved the following parameters from the oamWebServer.properties to wizard.properties. Refer to the following parameter mapping table.

oamWebServer.properties	**wizard.properties**
netborder.oamWebServer.defaultUDPListenPort	netborder.oamWebServer.sip.udp.listenPort
netborder.oamWebServer.defaultTCPListenPort	netborder.oamWebServer.sip.tcp.listenPort
netborder.oamWebServer.defaultTLSListenPort	netborder.oamWebServer.sip.tls.listenPort
netborder.oamWebServer.defaultUDPTCPSendPort	netborder.oamWebServer.sip.primarySipServerUDPTCPPort
netborder.oamWebServer.defaultTLSSendPort	netborder.oamWebServer.sip.primarySipServerUDPTCPPort

- Fixed bugs
  - Bug 6562 - Default audio law for BRI physical configurations should be A-law

- Bug 6575 - Error when updating from 4.1.0 to 4.1.3 with BRI config
- Bug 6577 - GW version is not displayed in the WEB UI
- Bug 6585 - Failed to update from NBE 4.1.2 to 4.1.3 under Linux

3.26 Release 4.1.3 Maintenance Release

- **New features**
  - Added support of ISDN-BRI multpoint
  - Added mechanism to configure gateway for a provisioning file generated by IPBX software such as 3CX
- **Know Limitations**
  - Status of the BRI channels remains in the IDLE state even if the cable is disconnected
  - Can't generate in-band progress tones (example: ring back) toward the PSTN for inbound CAS-R2 calls. Make sure the Telco switch is configured to generate progress tones.
  - Acoustic echo cancellation is not working properly. Please don't used it.
- **Configuration parameters changes**
  - PSTN configuration
    - Removed **TEI** ISDN-BRI configuration from the physical configuration
    - Added new ISDN-BRI call control configuration parameters: **topology**, **TEI** and **multipleSubscriberNumbers**.
- **Fixed bugs**
  - Bug 6489 – Can't detect reliably dial tone in some countries (example: Mexico).
  - Bug 6492 – Detection of the ringing signal coming on FXS is too long in some countries (example: Egypt)
  - Bug 6510 – Gateway may crash under certain conditions with ISDN-BRI cards.
  - Bug 6547 - High usage of SNMP & phone causes internal errors
  - Bug 6562 - Default audio law for BRI physical configurations should be A-law
  - Bug 6565 - Gateway does not always start with default license after update

3.27 Release 4.1.2 Maintenance Release

- **New feature**
  - Added support for Microsoft IE 9.0 and Firefox 4.0
  - Modified channel hunting algorithm to use digital channels prior to analog channels when the hunting is configured to REVERSE_LINEAR.
- **Know Limitations**
  - Bug 6489 – NBE does not reliably detect Mexico dial tone
  - Bug 6491 - INVITE with replaces scenario with no SDP in the invite seems to be broken
  - Bug 6500 -R2 no local ringback generated towards PSTN on inbound calls
- **Fixed bugs**

- Bug 6443 - NBE SIP UAS bind on 0.0.0.0 for UDP transport even if we specify a single IP address
- Bug 6462 - NBE should send CANCEL after SIP INVITE Timeout
- Bug 6467 - NBE should support SDP "a=silenceSupp:on" for CN support
- Bug 6468 -SNMP gwCallSource is invalid for J1 calls
- Bug 6473: FXS/FXO bad audio when using 10ms audio packets between the application and the driver
- Bug 6474 - Unreliable bringup on T1 lines, some enter alarmed state on gateway restart
- Bug 6480 - Smart update issue with file paths
- Bug 6483 – Mexico tone definition does not contains the SIT tone definition
- Bug 6486 - GW is not processing presentationIndicator and screeningIndicator on restricted J1 calls.
- Bug 6492 – FXO Ring Detection in Egypt

3.28 Release 4.1.1 Maintenance Release

- **Fixed bugs**
  - Fixed DTMF detector to avoid sending of RFC2833 events toward the SIP side due to residual echo present in the signal coming from the PSTN when a DTMF signal transmitted simultaneously toward PSTN.

3.29 Release 4.1.0 General Availability

- **New features**
  - Ability to export and import configuration via the WEB UI (version 4.1.0 and newer).
  - Ready to support rpm update
- **Know Limitations**
  - Bug 5345 – DTMF echo could be perceived when DTMF digits are transmitted toward an FXO interfaces even if the interface has been calibrated.
- **Fixed bugs**
  - Fixed various CAS R2 issues
  - Fixed various SIP protocol issues to pass more SIP torture tests
  - Bug 6048: FXO Calibration buttons clipped could be clipped when the WEB UI is used in light-box
  - Bug 6272: Quiesce-In Service is not working when the for 5ESS interfaces when the remote ends does not support SERVICE ISDN messages.
  - Bug 6283: SNMP Analog channels report dsx0Ds0ChannelNumber as 0 instead of 1.
  - Bug 6284: SNMP gwCallStats MIB: NoAnswer statistics is not counting unanswered calls properly
  - Bug 6285: SNMP gwCallStats MIB is not reporting the gwCallStartTime properly in the gwCallDetailsTable.
  - Bug 6424: RFC2388 events send by Linksys and Cisco phones are not interpreted properly because the marker bit is set on every packets.

3.30 Release 4.0.2 Limited Availability

- **New features**
  - Support for Linux CentOS 5.5 32 bit
  - Hardware
    - B600 Analog Voice Card – 4 ports FXO and 1 port FXS with hardware echo cancellation, PCI and PCI-Express
    - A500 ISDN BRI Voice Card – up to 8 ports with hardware echo cancellation, PCI and PCI-Express
    - B700 FlexBRI Hybrid Voice Card – up to 4 ports ISND-BRI and 2 Analog FXO or FXS ports with hardware echo cancellation, PCI and PCI-Express
  - PSTN
    - ISDN CAS R2 protocol (Central and South America countries only) terminal and network sides
    - ISDN-BRI terminal and network sides
    - FXO line calibration
  - SIP
    - HTTP Digest Authentication for INVITE and REGISTER methods
  - T.38 FAX relay
- **Know Limitations**
  - Bug 5264 : 401 and 407 messages are not recognized on invalid INVITE digest authentication
  - Bug 5268 : The gateway logs an ERROR at un-registration time when the last REGISTER was not answered
  - Bug 5449 : Bad behavior in Add/Remove boards in the PSTN configuration editor when removing a board and adding another if a FXS phone number is used in both boards.
  - Bug 5521 : AOC not supported in BRI
  - Bug 5642 : Call collisions (glare) not handled properly in R2
  - Bug 5649 : R2 logs statement are not correlated to call-id
  - Bug 5707 : In E1-R2, toggling from quiesce to in-service puts the gw in invalid state
  - Bug 5720 : SNMP is not supported for E1-R2 and BRI interfaces
  - Bug 5752 : R2 outbound calls failing with "handle_protocol_error" reported in SIP as 500 Internal Server Error
- **Fixed bugs**
  - Fixed various SIP interface issues to pass more SIP torture tests

4 Limitations and Known Problems

Here is the list of known problems and limitations.

4.1 Hardware & driver related limitations

- **AFT-400 analog telephony boards are NOT supported (bug 2235 & not tested).**
- **Support Sangoma Software version 7.0.2.0.** The gateway has been tested and validated with Sangoma software 7.0.2.0. The gateway validates this version and generates a warning if the version is different.
- **Echo cancellation tail length is fixed to 128ms for all calls.** The echo cancellation is performed by the hardware. Thus, having support for shorter tail length will have no impact of the overall performance of the gateway.
- Microsoft Windows Vista is NOT supported

4.2 Other limitations

- Bug 5720 : SNMP is not supported for E1-R2 and BRI interfaces
- Bug 5264 : 401 and 407 messages are not recognized on invalid INVITE digest authentication
- Bug 5268 : The gateway logs an ERROR at un-registration time when the last REGISTER was not answered
- Bug 5521 : AOC not supported in BRI
- Bug 5449 : Bad behavior in Add/Remove boards in the PSTN configuration editor when removing a board and adding another if a FXS phone number is used in both boards.
- Bug 4177 : When adding FXS interfaces in hardware detection mode through the PSTN Config tab in the WebUI, channel groups corresponding to the phone number of those interfaces are not automatically created. Hence, they have to be added manually for the gateway to be able to route calls to those interfaces.
- The gateway reports errors and fails to start when A200 boards with FXS modules are not connected to the power supply.
- Bug 3453 : DNS server list is interface dependent. The gateway does not send DNS requests to the right DNS server (primary) on a multi homed server when various interfaces are configured with different DNS servers. When all interfaces have the same DNS configuration, the right DNS server is used. There is a workaround for this bug.
- The gateway consumes 100% of the CPU when the server's clock is modified (Bug 3505).
- Analog PSTN interfaces seem to apply a gain on the output signal (bug 3272)
- Receiving a RE-INVITE does not force a re-keying in SRTP (bug 3242)
- International Telephony Tone Detection
  - Busy is erroneously detected as reorder or vice-versa in the following countries : Austria, Spain, Bulgaria, Denmark
  - Incorrect detection of several tones in the following countries : Jordan, Hungary, Philippines, South Korea, China
- Audio Quality. RTP Forward Error Correction (FEC) is not supported
- Wanpipemon tool
  - The gateway must be restarted after BERT tests are run for the gateway to be able to place calls on the tested channels.

  - The echo canceller must be disabled before BERT tests can be run
- **Documentation**
  - Included : Quick Start and Tone Configuration guides only
  - The Gateway User Guide is included in this distribution but is out-of-date, in particular the analog telephony support.
- **Analog disconnect supervision**
  - Low amplitude telephony tone are not always detected (Bug 1601)
- **Gateway Web User Interface**
  - Supported web browsers : Internet Explorer, Mozilla Firefox.
- **FXO caller-ID**
  - Support is limited to caller-ID extraction as described by Bellcore FSK 1200bps Caller-ID standards in SDMF or MDMF which is used in Australia, Canada, China, Hong Kong, New Zealand, Singapore and USA. The gateway extracts only the caller number from the caller-ID in SDMF mode and extract caller number and caller name in MDMF mode. ETSI FSK caller ID and caller name is also supported
- **Channels stay stuck in the Out-of-service state when we connect/disconnect a span in the out-of-service state. (Bug 2661)**
- **Servers with multiples NICs/IP addresses.** RTP traffic can only flow from a single, user-defined IP address (bug 1917).
- **Service shutdown while waiting to register/unregister to a SIP registrar may cause shutdown timeout:** If the feature of registering the gateway with a SIP registrar is used, and the gateway is waiting for a reply from a registrar that is particularly slow or down, it is possible that a service shutdown request times out in Windows before we can complete the operation (register or unregister). The impact is simply that the service shutdown is not very elegant.
- **Gateway Does Not Monitor the Via or Max hops Headers for Self-Loops:** If users design ill-formed routing rules, it could happen that they re-direct incoming SIP calls to the gateway's SIP user agent. The gateway does not currently ensure that the 'via' header is different from the source of the call nor that 'maxhops' is not violated. This could cause an infinite loop of SIP calls.
- **Limitations to the use of arbitrary SIP headers in the routing rules:**
  - If two headers of the same name are specified in the sip.out.header out parameters, only the last one is used
  - If a "known" SIP header (automatically generated by the gateway, as described in a point below) is used in sip.out.header, the header internally generated will not be overridden, creating two headers that have a great chance of confusing the remote SIP user agent.
  - Known SIP headers, automatically generated by the gateway, cannot be used as sip.in.header.* parameters. The list of all known headers follows:

```
VIA,
FROM,
TO,
CSEQ,
CALLID,
CONTENTLENGTH,
ACCEPTENCODING,
```

```
ACCEPT,
ACCEPTLANGUAGE,
ALERTINFO,
ALLOW,
ALLOWEVENTS,
AUTHENTICATE,
AUTHENTICATIONINFO,
AUTHORIZATION,
CALLINFO,
CCDIVERSION,
CONTACT,
CONTENTDISPOSITION,
CONTENTENCODING,
CONTENTTYPE,
DATE,
ENCRYPTION,
ERRORINFO,
EVENT,
EXPIRES,
HIDE,
INREPLYTO,
MAXFORWARDS,
MIMEVERSION,
MINEXPIRES,
MINSE,
ORGANIZATION,
PRIORITY,
PROXYAUTHENTICATE,
PROXYAUTHORIZATION,
PROXYREQUIRE,
RACK,
RSEQ,
RECORDROUTE,
REFERTO,
REFERREDBY,
REPLACES,
REQUIRE,
RESPONSEKEY,
RETRYAFTER,
ROUTE,
SERVER,
SESSIONEXPIRES,
SESSION,
SUBJECT,
SUBSCRIBESTATE,
SUPPORTED,
TIMESTAMP,
UNKNOWN,
UNSUPPORTED,
USERAGENT,
WWWAUTHENTICATE,
WARNING.
```