Sangoma Technologies Corp.

# Netborder Express Gateway for Linux

## Release Notes

V2.0 General Availability

September 11, 2009

# Netborder Express Gateway Release Notes

## Product Compatibility

Here are some of the major compatibility points.

- Operating Systems Supported:
    - Linux 32-bit CentOS 5 with the standard or PAE kernel (not Xen kernel)
- CentOS packages pre-requisites:
    - ncurses, ncurses-devel, libtermcap, libtermcap-devel, bison, libtool, flex, gcc, gcc-c++, automake, autoconf, imake, kernel and kernel-devel
- Sangoma Telephony Cards Supported:
    - AFT A101/2/4/8 T1/E1 with hardware echo cancellation (PCI / PCI-Express)
    - AFT A200 FXO with hardware echo cancellation (PCI / PCI-Express)
- Sangoma Software Release Versions supported:
    - 3.5.5 (included in gateway software package)
- SIP 3261 compliant endpoints using either TCP or UDP as the transport protocol
- DTMF relay as per IETF RFC 2833.
- RTP/RTCP as per IETF RFC 3550/3551
- Minimum Server requirements: Intel Core Duo 2 processor or later with a minimum 512 MB 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 analog interface<br>Support ISDN-PRI Q931 (DMS100, 4ESS, 5ESS, National ISDN 2) terminal and network sides.<br>NFAS for DMA100, 4ESS, 5ESS and National ISDN 2 variants terminal and network sides.<br>FXS Analog is not supported<br>NFAS with D-Channel backup is not supported<br>CAS 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 |
| ***RTP processing as per RFC 3550 and RTCP as per RFC 3551*** | G.711 codecs (uLaw and A-law) with law conversion. |
| ***DTMF per RFC 2833*** | Both DTMF relay (PSTN to SIP) and DTMF re-generation (SIP to PSTN) |
| ***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 a Linux Service*** | |
| ***Integrated to Linux init scripts system*** | |
| ***Configurable logging per sub-system*** | |
| ***Call logs*** | Per call information |
| ***Web Service Interface for management*** | |

## 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.

The gateway is licensed on a per telephony port basis. The license is host locked. To get a full license, simply get the MAC address of the server and e-mail it to Sangoma Technologies Corp. or to its duly authorized distributor with the number of ports required:

- support@sangoma.com

To get the physical address of the Ethernet adapter, simply start a Linux command prompt (ie: gnome-terminal) and execute the following command: “ifconfig -a”. Then look for the HWAddr item. It would look something like: 00:0B:DB:D8:06:00.

Please consult the user guide for more details.

## Limitations and Known Problems

Here is the list of known problems and limitations.

## Hardware & driver related limitations

- **AFT-400 boards are NOT supported.**
- **The gateway does not support T.38 Fax relay.**
- **Support Sangoma Wanpipe Software version 3.3.5.** The gateway has been tested and validated with Sangoma Wanpipe software 3.3.5. The gateway validates this version and generates an error if the version is different. The gateway will also fail to start.
- **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.
- **AMD processors are NOT supported.**

## Other limitations

- **Documentation (User Guide)**
  - Include Quick Start and Tone Configuration guides only
- **Analog disconnect supervision**
  - Low amplitude telephony tone are not always detected (Bug 1601)
- **Gateway Web User Interface**
  - Supported web browsers : Internet Explorer, Mozilla Firefox. Google Chrome is not yet supported
- **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
- **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.
- **Service shutdown may not be clean if there is activity in the gateway:** If the gateway is shutdown while there are active calls or is just starting, there may be errors in the log file. The errors may be safely ignored.
- **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.
```

## Changes Since Last Release

### Release 2.0 General Availability

- This software supports both analog FXO & digital PRI telephony interfaces.
- In band tone detection is now performed when required in PRI outbound calls.
- Added support of ISDN Network side (act like a telco switch) for all ISDN variants supported by the gateway.
- Added caller name support for all ISDN variants.
- Improved Web User Interface (UI)
  - All gateway configuration files can be edited through the UI

### Release 2.0 Limited Availability

- Extended of FXO connectivity to support most countries around the world (please consult the Web User Interface to get the list of these countries). However, the user has to edit tone definition files and the .RAM files used to regenerate call progress tones for all countries except AUSTRIALIA, CANADA and USA. Consult the Tone_Configuration_guide.pdf for more details.
- Added support to set the Type Of Service (TOS) field in the IP header of the RTP and RTCP packets transmitted by the gateway via the parameter “Netborder.media.ip.tos” in the gw.properties file.
- Extended FXO disconnect supervision to support battery removal and reverse battery disconnect detectors.
- Improved audio quality.

### Release 2.0 Beta

- This release is the first to offer FXO analog PSTN connectivity limited to North America countries.
- The Gateway Web User Interface has been redesigned and its capabilities have been greatly augmented :
  - The Gateway service can be started/stopped from the Web User Interface
  - Initial gateway configuration can be generated by a Web UI wizard
  - Most telephony configuration parameters can be modified through the Web UI.

## Release 1.6.2

- Fixed problem where the Windows user interface was not responding to user input as soon as the user had started the gateway WEB interface on server where the gateway is running. This problem was observed only systems having a single core CPU.

## Release 1.6.1

- Fixed a problem with the DTMF detection on B-Channel 23 of T1 spans.
- Fixed gateway crash that could occurs with some NFAS configurations.

## Release 1.6.0

- Fixed an issue where the gateway was not able to receive or make outbound calls with some ISDN switches configured in National ISDN 2 (NI2). The problem was caused by unexpected information element contained in the RESTART ISDN messages produced by the gateway.
- Modified the default install directory.

## Release 1.5.4

- Fixed an issue where the ISDN not able to establish the link when for ISDN group configured in NFAS with the PRIMARY span configured on the span different than the first span.
- Added a PSTN configuration ISDN parameter to disable or enable the initiation of the ISDN restart procedure when layer is coming up. This parameter (initiateRestartProcedure) is configurable per ISDN group. Please consult user guide for more details.
- Added new PSTN configuration ISDN parameters to control the behavior of the in-band progress tone generation (inBandProgressTonesGeneration) and the behavior of in-band progress tones indication (inBandProgressTonesIndicator). Please consult user guide for more details.
- Modified the gateway uninstall application to continue best- effort gateway removal process even if some errors occurs during the removal process. By doing this, the user can always uninstall the gateway.

## Release 1.5.3

- Fixed issue in E1 configuration where the ISDN layer 2 was not able to establish the link when the echo cancellation was enabled.

## Release 1.5.2

- Fixed RFC2833 multiple events for the same DTMF. Now consecutive events related to the same DTMF all use the same timestamp, as required by the RFC2833 specification. (Issue #1428)
- Fixed Error reporting on SIP message received with no 'Max Forward' header (Issue #1423).
- Fixed SIP messages SDP body so it ends with “\r\n”. (Issue #1427)

## Release 1.5.1

- Fixed SIP stack behavior to process multiple 2xx responses as described in section 13.2.2.4 of RFC3261. The gateway establishes the session with the remote user agent described by the first received 2xx response and rejects (ACK and BYE) the sessions described by any other 2xx response received after that.
- Added a new configuration parameter to insert or not the "rport" parameter in the Via header of the SIP requests sent by the gateway. For more details, please consult the parameter *"Netborder.sip.includeRportParameterInViaHeader"* in the Appendix B of the user guide.

## Release 1.5.0

- Now support incoming SIP messages with Date header field with time zone offset +/-HHMM to accommodate third party user agents that are not compliant to the restrictions imposed by RFC-3261 with regard to the Date header field.
- Added support of NFAS for DMS100, 5ESS and National ISDN 2 variants.
- Modified the gateway installation program to pre-install Sangoma device drivers.
- Modified the SIP registration configuration file to specify the SIP transport to be used to reach the registrar and the registered contacts.

## Release 1.1.3

- Fixed parsing of the transport parameter contained in the URL of the Contact .header for all SIP messages.
- Now support sending 183 Session Progress on INVITE without SDP body.
- Added support of the Q.931 RESTART primitive.
- Fixed SIP to SIP redirect handling.

## Release 1.1.2

- Fixed issue where the gateway failed to start because the gateway can't enable some Sangoma wanpipe devices after it reconfigure them. This behavior was observed on Windows 2003 server where there is one or more uninstalled third party devices.

## Release 1.1.1

- Fixed issue where the gateway may looses one or more span over time. This issue was observed only when the gateway was running with 8 spans or more.
- Fixed issue where the gateway WEB user interface sometime failed to display channel statistics when the gateway is heavily loaded.
- Fixed various warning messages about unexpected events that could happen when the gateway is heavily loaded.

## Release 1.1.0

- Added support for configurable voice quality enhancement features such as: acoustic echo cancellation, automatic level control, adaptive noise reduction and DTMF removal.

- Added support for a new echo cancellation mode optimized for speech recognizer. Also added configuration parameters for tail displacement, double talk behavior and comfort noise generation.
- Added support for E1 interfaces and ISDN-PRI Euro-ISDN (NET5, ETSI) variant.
- Fixed issue 1059: Added support of different number of physical interface and Isdn groups.
- Fixed issue 1013: Added configuration parameter “media.rtp.disableRtcp” to enable/disable RTCP on RTP port+1. When disabled, the port is still openned but received RTCP packets are dropped and none are sent.
- Fixed issue 955: Added support of B-Channel negotiation via the parameter “isdn/groups/group/BchannelNegotiation” in file pstn-config.xml.
- Fixed issue 946: Added support of OAM operations (Quiesce, In-service) on channels.
- Fixed issue 980: The message “ERROR - SangomaSpan::processDChannels> Sangoma::readMsg() failed (TIMEOUT)" does not appears in the logs anymore.
- Fixed issue 983: System performance does not degrade anymore when the gateway is restarted multiple times.

## Release 1.0.2 (alpha)

- Modified the product name. This invalidates all licenses issued for the previous releases. Please send an email to support@sangoma.com to get a valid license for this new version of the gateway.
- Fixed issue 985: Added support of early media.
- Fixed issue 1042: Added support of SIP Re-invites.
- Fixed issue 1053: Added support of SIP REFER.
- Fixed issue 1055: Added support of call progress tone generation such as ring-back, busy, rorder and SIT tones when the gateway receive a SETUP message that is not end-to-end ISDN.
- Fixed issue 1061: The size the RTP packets (specified in file “gw.properties” parameters “Netborder.media.rtp.packetSizeMs”) no longer need to follow the size of the Sangoma packets (specfied in file “pstn-config.xml for all sangoma interfaces via the attribute “voicePacketLengthInMs” of the BChannels sub element). However, voicePacketLengthInMs should be a common divisor of the all RTP packets. For example, set “voicePacketLengthInMs” to 10 ms to support 10, 20 and 30 ms RTP packets. If the gateway serves only 30ms RTP streams, set “voicePacketLengthInMs” to 30ms to get minimize the CPU usage and support more channels on the same workstation.
- Fixed issue 1067: The gateway reconfigures and restarts the WANPIPEs interfaces during the Gateway boot-up process.
- Fixed issue 1113: The gateway no longer crashes after several calls when it is connected to an Asterisk PBX.

## Release 1.0.1 (alpha)

- Fixed issue 1067: The gateway is not limited anymore to RTP packets of 20 msec.

## Release 1.0.0 (alpha)

- First version