

Application Note

Product: Ascom FreeNET VoIP Gateway
Purpose: Call Detail Records Configuration Guide
Date: January 24th, 2007

Introduction

This document provides an overview of the VoIP Gateway Call Detail Record (CDR) function. A CDR provides two main functions:

- Facilitate billing of customer calls performed through the gateway
- Gain insight into usage of telephony resources.

A CDR is defined as a sequence of entries which describes a single call. Calls that traverse the PRI interfaces will generate a CDR. Internal calls that do not traverse the PRI interfaces are currently not recorded.

The steps, screen shots, configuration, etc. depicted throughout this document are based upon VoIP Gateway software v1.2.1.

How are CDRs exported by the VoIP Gateway (IGWP)?

The CDR log entries are exported using a dedicated channel. The transport mechanisms are as follows:

- **syslog:** deliver entries to (unix-ish) syslogd.
Entries are sent to a configured syslog daemon using the syslog class specified.
- **raw tcp:** deliver entries to specified host and port in raw text-based tcp stream.
The IGWP will open a TCP socket to the specified IP address and port and write each entry to this stream. If the stream is closed for any reason, it will be re-opened again. Entries are separated by CR/NL.
- **http:** deliver entries by HTTP 1.0 GET request to specified web server with query part in URL (e.g. `http://path?log=msg`)

Can CDRs get lost?

Yes. Currently, CDRs are NOT permanently buffered. That is, when a CDR is generated and there is no active and functional CDR delivery mechanism at this point in time, the CDR gets lost. Thus, you must make sure that the mechanism is always in place.

More specifically, CDRs delivered by either raw tcp or syslog are always lost if the mechanism is not in place once the CDR is written. However, CDRs delivered via HTTP are buffered as far as memory suffices in the gateway.

The IGWP buffers up to 300kB for CDR records. A CDR is of variable size, but as a rule of thumb, 200 bytes per CDR should be a valid average.

How many CDRs are generated per call and when?

For diagnostic purposes CDRs can be used in a "verbose" mode where you will see a single CDR for each event on the call.

Here are the events for a typical straight call:

```
event=A:Call src_cgpn=32 src_cdpn=7300961321
```

```
event=B:Call src_cgpn=32 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
```

```
event=B:Call src_cgpn=32 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
```

```
event=B:Proceed src_cgpn=32 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
```

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```
event=B:Alert src_cgpn=32 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
event=B:Connect src_cgpn=32 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
event=B:Disc src_cgpn=32 cause=02_80_90 src_cdpn=7300961321 dst_cdpn=7300961321
dst_cgpn=32
event=A:Rel src_cgpn=32 cause=02_80_90 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
event=B:Rel src_cgpn=32 cause=02_80_90 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
```

For logging and/or billing purposes, you should configure the gateway to send “billing CDRs only” and you will only get a CDR generated when the call ends:

```
event=A:Rel src_cgpn=32 cause=02_80_90 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
event=B:Rel src_cgpn=32 cause=02_80_90 src_cdpn=7300961321 dst_cdpn=7300961321 dst_cgpn=32
```

Note: The real CDR will have much more information, including “connect” and “disconnect” times, from which talk time can be computed and billed.

CDRs and call legs

Each event in a CDR is prefixed by a **leg** identifier (either A or B). To understand this, consider the following example:

- An IGWP user (extension 2524) calls an external user in the PSTN (19192342567)
- The call is sent to the PSTN through an ISDN interface PRI1

When the call arrives at the IGWP (from the user), the GW1 interface mappings (if any) are evaluated. Then the call is sent to the routing engine. This end of the call is known as “**leg A**”. All events (CDR records) originating at this leg are prefixed with “A:”. Depending on the routing table configuration cdpn/cgpn (called party number/calling party number) mappings are performed, and then the call is routed to an outgoing interface. This end of the call is known as “**leg B**”. All events originating at this end are prefixed with “B:”.

In the CDR, you will find the numbers after the mapping at the source interface but before the mapping defined by the routing table as **src_cgpn** and **src_cdpn** respectively. The numbers after the mappings defined by the routing table but before performing the mappings defined for the destination interface are shown as **dst_cgpn** and **dst_cdpn** respectively. These numbers are always in sync on the A and B leg. However, please note that the numbers (especially the **dst_*** numbers) may change due to settings in the routing table (e.g. if least cost routing is configured and multiple call by call providers are tried to place the call). Also, the interfaces may change (e.g. if a trunk line bundle is used and one line is busy clearing the call with a “no b channel available” cause code, then another interface may be tried and thus seen as **dst_if**). Please note also that, after a call transfer, the **src_if** (which actually is the interface used by the A leg) may be used for an outgoing call (and the **dst_if** may be used for an incoming call), depending on the value of the **dir** attribute.

Initially, the A-leg is incoming (as seen from the routing engine), the B-leg is outgoing. However, after a call transfer, both legs may be incoming or both legs may be outgoing or the A-leg may be outgoing and the B-leg incoming. This may be significant for billing purposes. Each event has a “**dir**” attribute which is either “in” or “out”. So if there is an event for the A-leg and the “**dir**” attribute has the value “out”, then the call direction of the A-leg is outgoing (as seen from the routing engine).

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When a call transfer happens, it is reflected in an A:X or B:X event. If 2 calls are merged by a transfer, then one of the call legs receives an X event with an xref/xleg attribute which points to the call being merged.

All CDRs created by a single call, regardless of the leg the respective event originates from, share the same **ref** field (a unique reference number).

Depending on the billing strategy, the src/dst_dgpn (**diverting party number**) may be of interest.

How do I tell internal from external calls?

From the gateway process' point of view (and thus from the CDR's point of view), there is no distinction between internally originated and externally originated calls. Any call traveling through the gateway process is a call for which CDRs are generated.

As said before, internal calls between VOIP handsets currently do not create CDRs at all. For all others, the interpretation of being "internal" or "external" must be done by the CDR receiving application. Usually, such applications have a configuration dialog to define the interface names (which are seen as *dst_if* or *src_if*) which are considered "external".

How are CDR entries encoded?

CDR entries come in 8-bit ASCII character set.

CDR entries are URL-encoded. URL-encoding translates all reserved characters in a string to the format %HH, where HH is the hexadecimal equivalent of the character's ASCII value. Spaces are translated to "+". Example: If src_name is equal to "Mike O'Brien" then it will be encoded as "Mike+O%27Brien".

Fields are separated by ampersand ('&'). Each entry starts with a question mark. Example:
?event=A:Disc&time=24&ref=GW12:b5ed3f90e909d3119d37009033000048&src_cgpn=32&src_cdpn=10&dst_cgpn=32

There are slight variations depending on the delivery mechanism:

- syslog: a log class is added.
- raw-tcp: each CDR entry is prefixed by a time and date value. This is formatted as described above for the "date" tag (e.g. 19700101-000024).
- http: the CDR data is presented to the web server as FORM data which can be passed to CGI-interfaces, Active-X controls etc. Tag names are used as form-field names. Note that the CGI program receiving the data must return content to the web server to satisfy the GET command used. This content however usually is empty.

Fields in the CDR log entry

Each log entry has several fields. A field has a name tag and a value.

- Fields can appear in an arbitrary order within the entry.
- Not all fields appear in every log entry.

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Name tags include:

cause

Tag: cause

Example: cause=02_82_81

Cause for call termination. These are coded *<size>_<coding-standard/location>_<cause-code>[_<diagnostics>]*. The individual octets (separated by '_' are hex-numbers). *Size* is usually 0x2. *Coding-standard/location* is made up of: bit 8, 7, 6 and 5 usually 1000. Bit 4, 3, 2, 1 gives the location (that is, the entity generating the cause code). There may be extraneous octets giving further diagnostic information.

If bit 8 to 5 are unequal to 1000, the cause code is non-standard.

The location is usually 0010, so that the second octet is usually 0x82 (10000010).

The lower 7 bit of the third octet is the cause code. It is coded as follows (for standard codes):

Cause value	Cause
0x1	Unallocated number
0x2	No route to specified transit network
0x3	No route to destination
0x6	Channel unacceptable
0x7	Call awarded and being delivered in an established channel
0x10	Normal call clearing
0x11	User busy
0x12	No user responding
0x13	No answer from user (user alerted)
0x15	Call rejected
0x16	Number changed
0x1A	Non-selected user clearing
0x1B	Destination out of order
0x1C	Invalid number format
0x1D	Facility rejected
0x1E	Response to STATUS ENQUIRY
0x1F	Normal, unspecified
0x22	No circuit/channel available
0x26	Network out of order
0x29	Temporary failure
0x2A	Switching equipment congestion
0x2B	Access information discarded
0x2C	Requested circuit/channel not available
0x2D	Resources unavailable, unspecified
0x31	Quality of service unavailable
0x32	Requested facility not subscribed
0x39	Bearer capability not authorised
0x3A	Bearer capability not presently available
0x3F	Service or option not available, unspecified

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Cause value	Cause
0x41	Bearer capability not implemented
0x42	Channel type not implemented
0x45	Requested facility not implemented
0x46	Only restricted digital information bearer capability is available
0x4F	Service or option not implemented, unspecified
0x51	Invalid call reference value
0x52	Identified channel does not exist
0x53	A suspended call exists, but this call identity does not
0x54	Call identity in use
0x55	No call suspended
0x56	Call having the requested call identity has been cleared
0x58	Incompatible destination
0x5B	Invalid transit network selection
0x5F	Invalid message, unspecified
0x60	Mandatory information element missing
0x61	Message type non-existent or not implemented
0x62	Message not compatible with call state
0x63	Information element non-existent or not implemented
0x64	Invalid information element contents
0x65	Message not compatible with call state
0x66	Recovery on timer expiry
0x6F	Protocol error, unspecified
0x7F	Interworking, unspecified

The example cause cause=02_82_81 thus translates to “unallocated number”.

bcaps

Tag: bcaps

Example: bcaps=03_80_90_a3

Bearer capabilities as received in the incoming call. Coded as hexadecimal octet string according to Q.931. Octets are separated by ‘_’. Bearer capabilities are used to describe the type of calls (voice, fax, data, etc).

ref

Tag: ref

Example: ref=b8a9051be909d311b5fd009033000190

Call destination. This is a per-call unique identifier and shall not occur twice.

dst_cdpn

Tag: dst_cdpn

Example: dst_cdpn=01727250457

The called party number as dialed on the outgoing interface.

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dst_cgpn

Tag: dst_cgpn

Example: dst_cgpn=53

The calling party number as presented on the outgoing interface.

dst_dgpn

Tag: dst_dgpn

Example: dst_dgpn=70

The diverting party number as presented on the outgoing interface. This field occurs on calls that have been diverted after passing the gateway.

dst_name

Tag: dst_name

Example: dst_name=NVphone

The called H.323 alias name as presented on the outgoing interface.

dir

Tag: dir

Example: dir=out

The direction of the call-leg the event is posted for. E.g. if the record has an event=B:Connect and dir=out, then the direction of the B call leg is outgoing (as seen at the destination interface given as dst_if). Either in or out.

xref

Tag: xref

Example: xref=b8a9051be909d311b5fd009033000190

The ref of the call being merged into the current call due to a call transfer.

xleg

Tag: xleg

Example: xleg=A

The leg of the call referenced by xleg that is merged in. Either A or B.

event

Tag: event

Example: event=A:Call

The event triggering this particular CDR entry. Valid events are (more may be added in the future, so ignore unknowns!):

Eventname	Description
A:Call	The calling party initiates a call.
B:Call	The called party is called.
B:Proceed	A call proceeding is received from the calling parties network (or a transit).

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Eventname	Description
Media	A media channel has been established.
B:Alert	The called party is alerted.
B:Connect	The called party is connected.
A:X	The calling party transfers the call. Following events will display a different calling party.
B:X	The called party transfers the call. Following events will display a different called party.
A:Disc	The calling party disconnects the call.
B:Disc	The called party disconnects the call.
A:Rel	The calling party releases the call.
B:Rel	The called party releases the call.
No-Route	The gateway routing process has no route for the call.

src_cdpn

Tag: src_cdpn
Example: src_cdpn=01727250457
The called party number as received in the incoming call.

src_cgpn

Tag: src_cgpn
Example: src_cgpn=53
The calling party number of the calling party as received in the incoming call.

src_dgpn

Tag: src_dgpn
Example: src_dgpn=70
The diverting party number as received in the incoming call. This field occurs on calls that have been diverted before passing the gateway.

src_name

Tag: src_name
Example: src_name=Nvphone
The H.323 alias name of the calling party as received in the incoming call.

src_reg_name

ag: src_reg_name
Example: src_reg_name=Nvphone
The H.323 alias name of the calling party as defined by the gatekeeper.

src_if

Tag: src_if
Example: src_if=GW2
The originating interface of the call. E.g. GW2 can be a Gatekeeper interface.

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dst_if

Tag: dst_if

Example: dst_if=PPP

The destination interface of the call. E.g. PPP is a physical Isdn interface.

time

Tag: time

Example: time=95985

A time-stamp in 1 second resolution since 01.01.1970. The time base does not change during a call (it might change between calls due to clock settings).

alert_time

Tag: alert_time

Example: alert_time=95985

A time-stamp in 1 second resolution since 01.01.1970. This timestamp is comprised within a CONNECT CDR, in order to ease the duration calculation of the alerting phase. :

$\text{alert_duration} = \text{time}(\text{CONNECT}) - \text{alert_time}$

connect_time

Tag: connect_time

Example: connect_time=95985

A time-stamp in 1 second resolution since 01.01.1970. This timestamp is comprised within a DISC or REL CDR, in order to ease the duration calculation of the phase a call was in the connected state. :

$\text{connect_duration} = \text{time}(\text{DISC}) - \text{connect_time}$

disc_time

Tag: disc_time

Example: disc_time=95985

A time-stamp in 1 second resolution since 01.01.1970. This timestamp is comprised within a REL CDR, in order to ease the calculation of the amount of time it took to completely release a call.:

$\text{release_duration} = \text{time}(\text{REL}) - \text{disc_time}$

date

Tag: date

Example: date=20000101-000000

The "human readable" date and time: 4 digit year, 2 digit month, 2 digit day of month, 2 digit hours (24h clock), 2 digit minutes, 2 digit seconds. This is not implemented in beta III software.

xcoder

Tag: xcoder

Example: xcoder=G729AB,60c(0,0)

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The media codec being applied in transmit direction. If present, the info conveyed is:
Codec[string],Framesize[ms](Receive Jitter[ms],Roundtrip Delay[ms])

Currently, the following coders are defined:

G711A, G711u, G723-53, G723-63, G729, G729A, G729B, G729AB, G726-40, G726-32, G726-24, G726-16, T38TCP, T38UDP, XPARENT, HDLC¹

rcoder

Tag: rcoder

Example: rcoder=G729AB,60(0,0)

The media codec being applied in receive direction. If present, the info conveyed is:
Codec[string],Framesize[ms](Receive Jitter[ms],Roundtrip Delay[ms])

xstats

Tag: xstats

Example: xstats=0+0+0+0+0

The media statistics of the outgoing audio stream as measured on the remote party. The five fields are: Average Roundtrip Delay, Maximum Roundtrip Delay, Average Jitter, Maximum Jitter, Packet Loss.

rstats

Tag: rstats

Example: rstats=0+0+0+0+0

The media statistics of the received audio stream. The five fields are: Average Roundtrip Delay, Maximum Roundtrip Delay, Average Jitter, Maximum Jitter, Packet Loss.

srv_id

Tag: srv_id

Example: srv_id=00-90-33-01-00-7a

The server id. The purpose is to provide a constant unique id serving as an originator address for CDRs. The actual aim is to support billing applications.

Sample billing record and interpretation

The following example shows a raw Call Detail Record as captured by a syslog server.

```
?event=A:Rel&time=1169481480&date=20070122-155800&ref=1d03db78e909d31181f700013e00953b&dir=in&src_if=RP1&dstif=PRI1&src_cgpn=2204&src_cdpn=5201&src_name=Lab+IP+4&dst_cgpn=2204&dst_cdpn=5201&src_reg_name=EXTERN&bcaps=03_80_90_a3&cause=02_80_90&xcoder=G711u,30(30,0)&rcoder=G711u,30(743,23)&xstats=30+30+0+0+0&rstats=743+743+23+23+0&alert_time=1169481458&connect_time=1169481466&disc_time=1169481480&srv_id=00-90-33-08-55-00
```

The **most pertinent fields** (separated by &) are interpreted as:

?event=A:Rel A-party (the calling party) has disconnected

¹ Although HDLC will currently never happen

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date=20070122-155800 January, 22, 2007 15th hour, 58th minute, 00 seconds GMT
src_cgpn=2204 2204 is the A party (calling party), a user on the Gateway
src_cdpn=5201 5201 is the B party (called party), reached via the PRI
connect_time=1169481466 Time stamp at beginning of conversation (seconds since
1/1/1970)
disc_time=1169481480 Time stamp at end of conversation (seconds since
1/1/1970)

This data is processed to yield: Local user 2204 called external party 5201 at 10:58:00 AM, January 22, 2007 and talked for 14 seconds.

Contact information

For any additional questions, please contact the Ascom Technical Assistance Center at 1-877-71-ASCOM, option 3.