Call Accounting in a VoIP Infrastructure

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Abstract. *An H.323 CDR collecting and consolidating architecture based on Radius is presented. The important fields involved in the consolidation process for Cisco gateways and GnuGK CDRs are investigated and a unique record format is defined. From the database, various statistics are extracted including disconnection causes, call distribution over a day, number of calls over a day period, call quality along a day and number of simultaneous calls.*

1. Introduction

Data and voice networks are essential to any modern enterprise or organization. These networks have been historically deployed as independent infrastructures. Today, convergence towards a unique network capable of supporting both services can lead to saves, besides paving the road for rising new integrated multimedia applications.

As in traditional telephony systems, also in voice over IP (VoIP) networks there is a need for collecting call statistics, be for accounting, billing or quality assessment reasons. Billing can be based on different factors as origin/destination, call duration and time of day, among others. Through call analysis it is possible to verify if resources such as telephony trunks, voice ports and reserved bandwidth in data links are adequately provisioned for the service. In the case of VoIP, it is also possible to collect voice call quality information which can be used to identify network congestion problems, inadequate QoS configuration, insufficient resource provision, or the need to change parameters in traffic engineering. Furthermore, more sophisticated admission control strategies can be required in VoIP networks and voice statistics can be handy.

In telephony networks, messages used in call billing among providers follow standards defined by organizations as Telcordia, which in its document GR-1100 [1] specify BAF (Billing Automatic Message Accounting Format), an industry standard for record format and semantics. In VoIP networks there is no standard format, neither a specification of how these records have to be generated. Radius protocol, IETF standard for remote dial in remote authentication, was extended to cover delivery of accounting information. Some VoIP vendors are also using this protocol to delivery Voip Call Detail Records (CDRs) from gateways, H.323 gatekeepers and SIP servers to a Radius accounting server. SNMP is also used by some gateways by the use of private MIBs.

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Many institutions and network providers have developed their own accounting systems. Czech (TEN-155 CZ) [2] and Australian (AARNET) [3] academic and research networks have developed accounting and billing applications which use CDRs emitted through Radius. ECAS (*Enterprise Call Analysis System*), developed at National Chiao Tung University, Taiwan, also processes CDRs generated through Radius [4].

In Brazil, the Brazilian Education and Research Network (RNP) offers a VoIP service in its backbone, named fone@RNP[2]. To provide accounting and statistics for this service, an architecture was developed to allow the collection and recording of CDRs, besides the emission of performance and usage reports. Considering the heterogeneity of RNP network, we have also adopted Radius as the standard protocol for CDR collection.

Radius protocol defines syntax and data format, however it also permits inclusion of vendor specific data in the CDRs. In VoIP, the way CDRs are reported by Radius also varies according to the type of call and involved equipment.

A cautious analysis of involved equipment behavior and the way how the call was made is necessary to identify a correct consolidation procedure, where CDRs from different sources are merged in a unique call consolidated record. In the development of the collection architecture is important the definition of a standardized record for all types of calls, to allow greater performance in statistics processing of a large database. Furthermore, the correct consolidation of the various CDRs of a same call is essential to guarantee reliable statistical information.

Fone@RNP uses standard hierarchical E.164 PSTN addressing. Though SIP and H.323 are presently supported in fone@RNP, SIP has been only recently introduced and H.323 remains the only structure responsible for processing E.164 and the major VoIP protocol in use. Presently, accounting and call collection is done only for H.323 and the implemented H.323 accounting architecture is the focus of this work.

2. Using Radius in VoIP networks

RFC 2867 defines Radius as the accounting protocol for dial-up connections, and specifies the usable attributes in this scenario. Radius is also used for VoIP accounting. Twelve attributes (codes 40 to 51) where specifically defined for accounting purposes. Depending on the devices involved in a single H.323 call, many Radius records can be generated. The attribute *Acct-Session-ID* allows finding all correlated records.

Radius vendor-specific attribute (VSA) is important because it can be used by vendors and developers to report not standardized specific information. Cisco Systems has defined a set of VSAs specifically for VoIP accounting.

When a call is completed through a VoIP gateway, two call legs are established. When a second gateway is involved, a total of four call legs are associated to the call, two in each gateway. A CDR is emitted for each call leg and all records of a call are identified by the VSA *h323-conf-id* attribute, which should be the same in all related records.

Depending on call leg type, specific VSAs will be reported in Radius records. VSAs which contain parameters that give indications of the average quality of a call (*h323-voice-quality, late-packets, lost-packets, round-trip-delay* e *early-packets*) are the most important.

[2] Available in http://www.rnp.br/vo ip

At call completion, the gateway generates one record for each call leg. VSA *release-source* permits identify the segment which caused the end of a call. VSAs *h323-disconnect-cause* and *disconnect-text* in the related CDR identify the disconnection cause. Other VSAs, namely *h323-setup-time*, *h323-connect-time* and *h323-disconnect-time,* determine the time of occurrence of setup, connect and disconnect messages. VSAs *tx-duration* and *voice-tx-duration* indicate the duration of a call.

GnuGK is the software used for gatekeeper and it can be configured to emit CDRs with Radius protocol (feature available from version 2.0.5 up). Contrary to gateways, which associate many call legs to a call, for a call GnuGK controls, it emits only two CDRs, at beginning and end of a call, both having VSA *h323-call-origin=proxy.*

Two kinds of Radius packets are used in accounting: *accounting-request* (code 4) and *accounting-response* (code 5). The first type is used for a client to send call data to a Radius server. After storing the information, Radius answers with a packet of the second type as a confirmation. If this confirmation is not received, the client has to repeat the procedure to the same server or use an alternative server, depending on configuration. This mechanism is vital to guarantee a reliable operation with no data loss. However, this procedure can provoke record duplication. A*cct-status-type* and *acct-session-id* attributes can be used to identify and remove duplicated records.

Radius packets are authenticated in the server with passwords associated to each authorized Radius client.

3. Accounting and Statistics Architecture

CDRs generated by gateways and gatekeepers are collected via Radius and stored in a SQL database for processing. In our experimental set, FreeRadius version 0.9.3 was used as accounting server and MySQL version 4.0.17 was used for CDR storage. At present, PostgreSQL is the database in use because of better performance and availability of additional features compared to MySQL.

FreeRadius regular code is not able to record these VSAs in MySQL. As some of these VSAs are important to call evaluation, the code in file *rlm_preprocess.c* was modified to allow recording. Also, *db_mysql.sql* script which holds the schema for the *RadiusAcct* database and the Cisco dictionary, used by FreeRadius to identify the VSAs, had to be modified[3].

The different CDRs associated with a same call have to be analyzed to generate single consolidated information. This procedure is initiated as soon as a call completion CDR is received. Then, all CDRs related to the call, which are stored in temporary SQL tables, are analyzed and one consolidated record is produced.

Table 1 – Consolidated record fields

Objective	Information	Where attributes are obtained from
Involved users	Calling E.164	CallingStationId
	Called E.164	CalledStationId
	Calling IP	FramedIPAddress in GK records (h323_call_origin = Proxy)
	Calling party type of phone	In records with h323_call_origin = answer: h323_call_type = voip → VoIP client

[3] Code is available in http://www.voip.nce.ufrj.br/download_pt.htm

Objective	Information	Where attributes are obtained from
		h323_call_type = telephony → extension, Cisco IP phone or PSTN
	Called party type of phone	In records with h323_call_origin = originate: h323_call_type = voip → VoIP client h323_call_type = telephony → extension, Cisco IP phone or PSTN
Call characteristics	Call identification	h323_conf_id
	Call Answered?	Not answered calls have blank h323_connect_time attribute in GK records (h323_call_origin = Proxy)
	Local call?	Local calls have remote_media_address = local GK IP in records where h323_call_origin = answer
	Normal clearing?	Normal end has h323_disconnect_cause = 10. If Release_source = 2 , check record with h323_call_origin = originate ; if Release_source = 3, check record with h323_call_origin = answer
	Disconnection cause	h323_disconnect_cause and Disconnect_text. If Release_source = 2 check record with h323_call_origin = originate, if Release_source = 3, check record with h323_call_origin = answer.
	User who initiated disconnection	If Release_source = 2, disconnected by called party. If Release_source = 3, disconnected by calling party.
	Call quality	h323_voice_quality in records with h323_call_type = VoIP
Involved Institutions	calling party institution	Analysis of calling GK IP
	Called party institution	Analysis of called GK IP
	Calling GK IP	NASIPAddress in record with h323_call_origin = Proxy
	Called GK IP	Remote_media_address in record with h323_call_origin = answer
Time information	Time_setup	h323_setup_time
	Time_connect	h323_connect_time
	Time_disconnect	h323_disconnect_time
	Duration	tx-duration

The use of a variety of clients (IP phones, ATAs and softphones), the type of call (between two VoIP clients, between a VoIP client and a PBX extension, between two PBX extensions, between a VoIP client and a public telephone), the kind of involved equipment (one or more gateways and gatekeepers), the type of call end (normal or abnormal) and who has ended the call (VoIP client, extension or public telephone) generate a variable number of different types of consolidated records. A standard consolidated record format to be applied to all types of call was defined. See Table 1.

A Web application was developed in PHP to allow flexible and easy plot of selected statistics. The following statistics can be obtained:

- Number of calls per day over a period of time (answered and unanswered).
- Number of calls per hour along a specific day (answered and unanswered).
- Minimum, average and maximum call duration per hour in a specific day (only answered calls).
- Disconnection cause.

  Evaluating the cause of a call disconnection (*h323-disconnect-cause* and *disconnect-text* attributes) allows the identification of problems with are causing abnormal call clearing. Information on the real cause of disconnection should be obtained in the call leg reported in *call-release-source*. It is important to point out that some calls which have ended normally are reported with abnormal clearing. For example, a VoIP gateway using FXO and identifying call disconnection through supervisory tones (*supervisory disconnect dualtone*), will receive "*User busy*" tone as a disconnection indication from the PBX and report this status as disconnect cause, even if the call has ended normally.

- Call quality

To help identify possible misconfiguration or malfunction that can be interfering with a VoIP service in a network it is essential to evaluate call quality. Cisco gateways report this information through *h323-voice-quality* attribute in VoIP call legs, which represents ICPIF (*impairment/calculated planning impairment factor*) parameter calculated according to ITU-T E-model (ITU-T G.107 [5] and G.113 [6] recommendations). This parameter takes into account type of codec, packet loss percentage and average delay. In our application, quality levels were defined according to ICPIF range values, as shown in Table 2.

Table 2 – Range for ICPIF index

ICPIF	*Call Quality*
0 – 5	Very good
6 – 10	Good
11 – 25	Regular
26 – 55	Bad

Different plots for visualizing call quality information have been implemented to help analyze the different aspects of VoIP service.

It is important to say that quality is measured in only one direction. Therefore, it is necessary to evaluate VoIP call legs in both directions to be able to estimate the quality experienced by the two speakers, what may require a gateway to terminate each end of a VoIP call.

In calls between a VoIP client and a traditional PSTN telephone, the gateway reports the quality received by the PSTN network, while no quality indication is available to estimate the received voice at the VoIP client. In a network, where peer to peer communication among VoIP clients is frequent and user service satisfaction is a major goal, lack of adequate call quality statistics may be a considerable impairment.

To circumvent this limitation, VQuality, a library implementing the E-model and proposed extensions [7] and also able to generate CDR with extensive quality information (called VQCDR), was developed. This library has been integrated to OpenPhone H.323 client to generate VQOpenPhone client. With the widespread use of this new client and the deployment of a quality monitoring architecture for collecting VQCDRs, a comprehensive call quality statistics can be obtained.

- Call intensity

To be able to evaluate if network VoIP resources are appropriately provisioned for the offered voice traffic, it is important to compute the average number of simultaneous calls per period of time along a day or traffic intensity.

4. Conclusions and Future Work

We have described the most important requisites for the development of a tool to evaluate statistics associated to calls in a heterogeneous H.323 VoIP network. Device diversity generates different call detailed records, from the point of view of type, behavior or meaning of stored information. A matrix describing the different behavior was elaborated to allow merging various CDRs of a same call into one consolidated record. With this consolidation procedure, statistics extraction can be done more efficiently and in shorter time. A great variety of statistics are obtained, specially disconnection cause, call distribution along a day, call distribution over many days, call quality along a day and number of simultaneous calls over periods of time, which is denominated call intensity.

The developed tool is focused on statistics collected in only one institution. We are developing a module to centrally store call consolidated information from all institutions. Global reports generation and more correlated analysis would then be possible to be produced.

We have analyzed only H.323 CDRs. SIP CDR analysis is on going and a comprehensive consolidation process will be implemented. In this particular, CDRs emitted by Asterisk, which acts as a gateway between H.323 and SIP, and CDRs emitted by SIP proxies are of fundamental importance. CDRs emitted by Asterisk working as VoIP gateway or softPBX will be integrated to the operational architecture.

5. References

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