OmniVista 8770

Accounting and VoIP Ticket Collectors

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1

Overview

1.1 Introduction

The Communication Server Call Detail Records (CDR) contain various data items (called number, call type, duration, ...) designed to allow call costs to be charged by an external accounting application.

The OmniVista 8770 application is able to:

- Connect to a Communication Server system periodically
- Retrieve Call Detail Records (CDR) and VoIP tickets by FTP transfer
- Copy the files that contain the CDRs and VoIP tickets on a disk available in the OmniVista 8770 server
- Store these files on disk for a defined period

The Communication Server VoIP Statistics records contain various data items (codec used, jitter impact, packet loss, delay...) designed to identify the causes of potential problems with IP devices connected to a Communication Server that could result in a bad Quality of Service (QoS).

The VoIP tickets for OXO Connect and OmniPCX Enterprise are similar in content but are different due to for example, hardware and network aspects.

The data stored in the collector can be used by the OmniVista 8770 to consult the:

- Communication Server taxation ticket from the OmniVista 8770 Accounting application
- OXO Connect and OmniPCX Enterprise VoIP tickets from the OmniVista 8770 Accounting application

The Collector data can be exported to third party applications to provide Communication Server Statistics records for analysis or accountancy.

In this document the term Communication Server can refer to a:

- OmniPCX Enterprise server
- OXO Connect server

1.1.1 Accounting tickets generated by Communication Servers

1.1.1.1 Tickets collected on OmniPCX Enterprise

The OmniPCX Enterprise generates and stores several *.DAT files containing all CDRs concerning the same hourly period. Filters must be applied for external accounting. On the Communication Server:

1. Each hour a command account compress is automatically launched to generate a compressed file TAXAXXXX.DAT containing one file in text format with same name TAXAXXXX.DAT.
2. The name of this file is added to ACCOUNT.LIS
3. ACCOUNT.LIS and .DAT are stored under usr4\ACCOUNT

Note 1:

XXXX is a counter using letters

TAXAxxxx.dat are stored on the Communication Server for 94 days

On the OmniVista 8770 when a synchronization request is submitted:

1. The command account compress is forced on the Communication Server
2. The ACCOUNT.lis is loaded
3. The information Last Polled File is read from Communication Server declaration
4. Only the file present in ACCOUNT.LIS and which counter are after the last polled file are retrieved via ftp
5. he information Last Polled File is updated

Note 2:

*If the accounting process is enabled, *.DAT files are directly loaded into OmniVista 8770 database*

*If the collector is enabled, the *.DAT file are uncompressed into the collector folder under a subdirectory named Networknumber=X\SubnetwornodeNumber=YYY*

1.1.1.2 Tickets collected on OXO Connect

OXO Connect works with a buffer of CDR tickets. On the Communication Server you have to specify some parameters for accounting

When a synchronization request is submitted by the OmniVista 8770,

1. The buffer is loaded as a unique file
2. A binary file is loaded and named ALZxxxx.ALZ
3. A header is added to this file with OXO Connect node number and version
4. The information Last Polled File is updated
5. On the Communication Server, the buffer content is reset

Note:

- *If urgent alarms from OXO Connect are retrieved by OmniVista 8770, a synchronization is forced on the Communication Server*
- `xxxx` *is a counter using number*
- *if accounting process is enabled,* `*.ALZ files` *are loaded directly into the OmniVista 8770 database*
- *If the collector is enabled, the* `*.ALZ` *file is uncompressed into the collector folder under a subdirectory named* `Networknumber=X\SubnetwornodeNumber=YYY`

1.1.1.3 Manual synchronization

In certain cases, recent data may be needed for immediate analysis. In this case a manual synchronization can be launched to collect the data.

1.1.2 VoIP tickets generated by Communication Servers

The Communication Server VoIP Statistics records contain various data items (codec used, jitter impact, packet loss, delay...) designed to identify the causes of potential problems with IP devices connected to a Communication Server that could result in a bad Quality of Service

(QoS).

The VoIP tickets for OXO Connect and OmniPCX Enterprise are similar in content but are different due to for example, hardware and network aspects.

An IP ticket is generated at the end of each call involving one of the following devices:

- IP Phones (Advanced e-Reflexes and IP Touch)
- IP board
- PCMM (Softphone) (OmniPCX Enterprise only)
- eVA (4645 Voice mail) (OmniPCX Enterprise only)

A call between two Alcatel-Lucent IP sets may be divided into one or more segments, each segment generating two tickets at the end-points of the segment.

For example, a call between an IP Phone A and an IP Phone B from the same node will create one segment with two tickets, one ticket generated from IP Phone A and another one from IP Phone B.

More complex cases can be handled such as a call between an IP Phone and a UA phone between two nodes.

Example cases for a complex call include:

- More than one segment can be created
- A segment dedicated to signalling (Ringback tone) could be created
- If one of the IP end-users is not an Alcatel-Lucent IP device, no ticket is generated by this end user and only one ticket will be generated (by the Alcatel-Lucent IP device) for this specific segment
- if domains are managed with codec restrictions, more IP segments could be created (intermediate paths through DSP board) in the case of extra-domain calls, ...

Note:

- *Tickets generated for a Ringback tone are not useful and must not be take into account for QoS analysis.*
- *Non Alcatel-Lucent IP devices do not generate IP tickets. For example, a segment between a GD board and an H323 terminal will generate only one ticket from the GD board (the H323 terminal does not generate IP ticket)*
- *PBXs can also generate CDR tickets (refer to Accounting tickets generated by Communication Servers) but these have no relation with IP tickets. No correlation can be achieved between the two types of ticket.*

1.1.3 Communication Server ticket generation

OmniPCX Enterprise generate on its disk, several compressed IP ticket files with extension ".DAT" containing all IP tickets concerning the same hourly period.

On the OmniPCX Enterprise

- Each hour the process "account compress" is automatically launched
  - This generates a compressed file IPXXXX.DAT containing one file on text format with same name IPXXXX.DAT. Compression algorithm used is LZW algorithm (this one could be decompressed through winzip, gzip, ...).
  - the name of this file is added to IP.LIS log file.

- IP.LIS and IPXXXX.DAT files are stored under \usr4\

Note:

- *XXXX is a counter using letters. The letters are incremented in the ascending order*
- *A “IPxxxx.dat” file is kept on OXE for 15 days by default*
- *Compressed VoIP statistic ticket file may be generated more frequently by issuing the command “account compress” on the OXE to generate them*
- *IP tickets generated on a PCS (when this is active) are named: **IPXXXX<PCS_IP_@>.DAT** where **<PCS_IP_@>** is the hexadecimal value of the PCS IP address*
- *On CPU duplication configuration IP ticket files generated.on main CPU are duplicated on Standby CPU. The same for PCS when this one is in inactive state.*

1.2 Communication Server connectivity requirements

The following table indicates:

- The release requirements for ticket collection with the OmniVista 8770
- The file extension used for the collected files
- Connectivity and protocol used

Communication Servers supported	Release supported	Collected Files	IP connectivity	Modem connectivity
OXO Connect	All release from R5.0	Binary ***.alz**	Yes	
OmniPCX Enterprise	All release from R8.0	Text ***.dat**	Yes	

1.3 External application requirements

External applications need to:

- Load IP Tickets files (IP*.DAI) from the collector directory
- Load IP Tickets files as binary files
- Use IP Tickets data to compute VoIP QoS

The third party application could interact with its database to answer to user requests:

1. The user defines the specific search criteria to produce ticket field values. The criteria are used define a filter
2. The application contacts the database and supplies the filter to apply to data.
3. The database returns a set of results to the application containing the tickets that match the filter.
4. .The application displays the resulting ticket list in thetickets overview.

It could be possible to define different types of filter:

- Connection related fields: local and distant IP address, network number, node number, crystal number, coupler number

- Call related fields: local and distant call reference, call duration, equipment type
- Quality related fields: RTP packet lost, jitter impact, delay, ...
- User defined filter: the user can defines
  - The fields
  - Apply value limitations
  - Apply an arithmetic operator

By using the IP ticket data, the application can calculate new QoS fields like MOS, jitter average, RTP packet lost percent, ...

1.4 License requirement

Specific Licenses are required for the OmniVista 8770 to collect the CDRs and VoIP tickets effectively.

You can verify the license status from the help menu of OmniVista 8770 application (access path: **Help** > **About**).

The list of possible licenses is:

- *Ticket collector* license used to access the ticket collector feature. Its value represents the maximum number of users that can be managed by the ticket collector (Both for CDR and VoIP tickets)
- *Accounting* license used to access the accounting application. Its value represents the maximum number of users that can be managed by the accounting application.It gives also access to the CDR ticket collector feature
- *Performance* license used to access Voice over IP performance observation application for OmniPCX Enterprise. Its value represents the maximum number of OmniPCX Enterprise users that can be managed by the VoIP observation application. It gives also access to the VoIP ticket collector feature from OmniPCX Enterprise
- *OXO Voice over IP Performance* license used to access Voice over IP performance observation application for OXO Connect. Its value represents the maximum number of OXO Connect users that can be managed by the VoIP observation application. It gives also access to the VoIP ticket collector feature from OXO Connect

Chapter

2

Configuration procedure

2.1 Prerequisites

To configure the OmniVista 8770 for ticket collection:

- The OmniVista 8770 and its applications must be installed correctly and operational
  For more information, refer to the OmniVista 8770 Installation Manual, see Installation - Overview
- The Communication Servers must be declared in the OmniVista 8770 **Configuration** application
  For more details about declaring Communication Servers, see Configuration - Operation - Declaring a Communication Server Network.

2.2 Configuring OmniPCX Enterprise parameters for VoIP ticket transmission

This refers to a system option in the OmniPCX Enterprise management tool, to activate or to deactivate IP ticket transmission.

1. Select IP
2. Enter the following attributes:

Attributes	Values
IP Ticket	Select and validate **Yes** to activate ticket transmission. Default value: **No**
Max Nb. Days of Storage	Enter the number of days for the storage of compressed files, from 1 to 31. Default value: **15**

3. Validate

Verify the following accounting parameter:

1. Select Applications > Accounting
2. Enter the following attributes:

Attributes	Values
Files for External Accounting	Select and validate **Yes** to enable ticket transmission to the external application. Default value: **No**

3. Validate

2.3 Configuring the ticket storage parameters

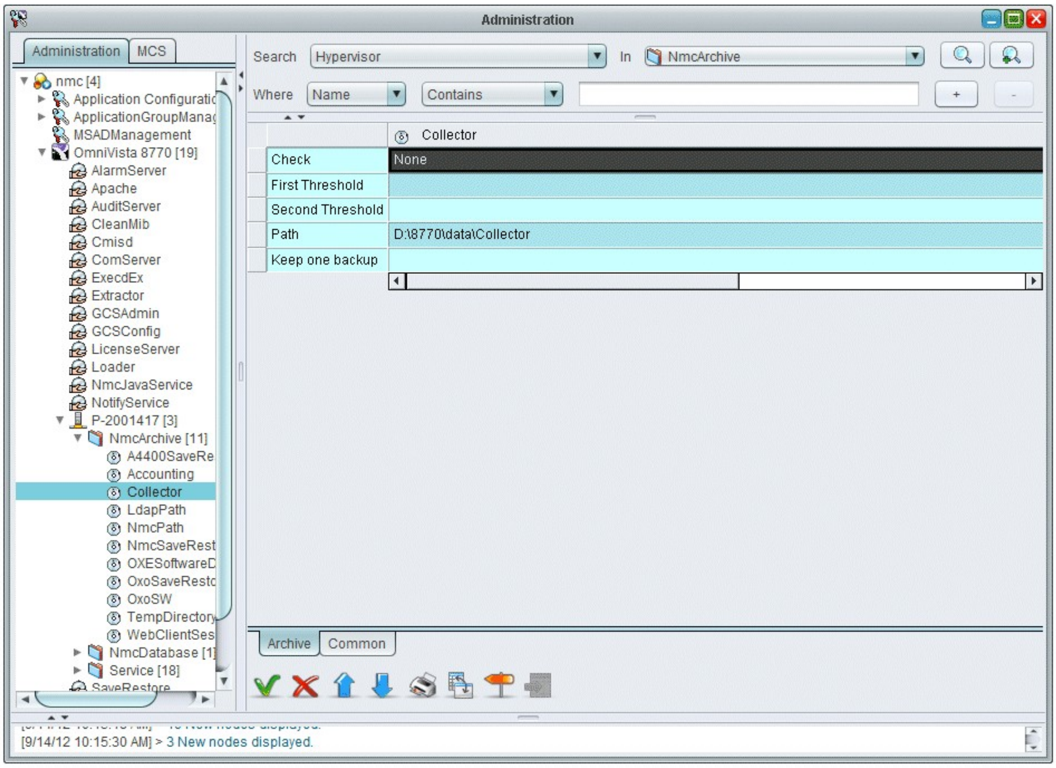
There are two collector parameters that you can modify:

- Path used to collect tickets

- Delay (number of days) that the collected data files are stored before being deleted

To modify the collector path:

1. From the OmniVista 8770 **Administration** application, select **nmc** > **OmniVista 8770** > `name of server` > **NmcArchive** > **Collector**



2. In the **Path** field, enter the path used to collect tickets
3. In the **Common** tab, enter the delay period for collected date to be stored in the **Clean-up Delay** field

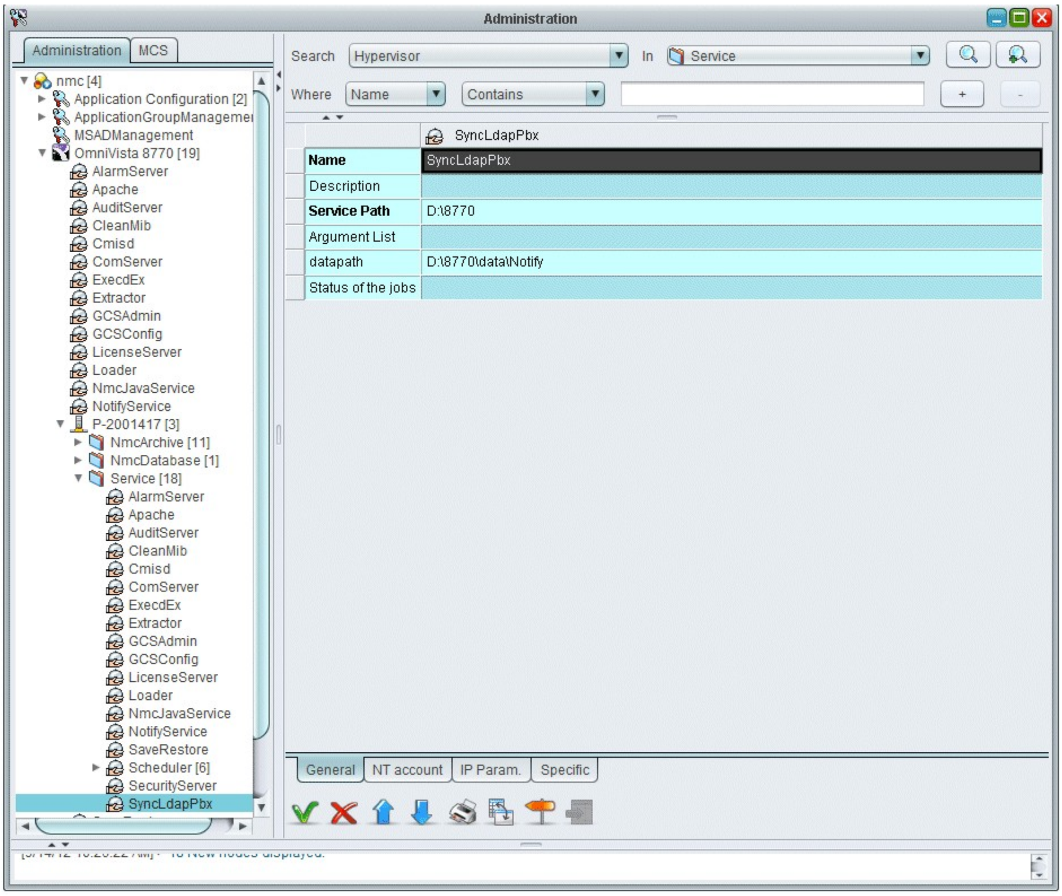
For Windows®

In case the path is not to a local drive but to a remote PC, you have to define a user that will be able to see this drive and add the following access rights for this user:

- Act as part of the operating system
- Increase quota
- Replace a process level token
- Open a session locally

On OmniVista 8770:

1. In the **Administration** application, select **nmc** > **OmniVista 8770** > name of server > **Service** > **SyncLdapPbx**



2. In the **NT account** tab, enter the **Username** and **Password**

   Note:
   NT username format is:

   - `Ntdomain\userlogin` *for login defined in a domain*
   - `Pcname\userlogin` *for login defined in a Workgroup*

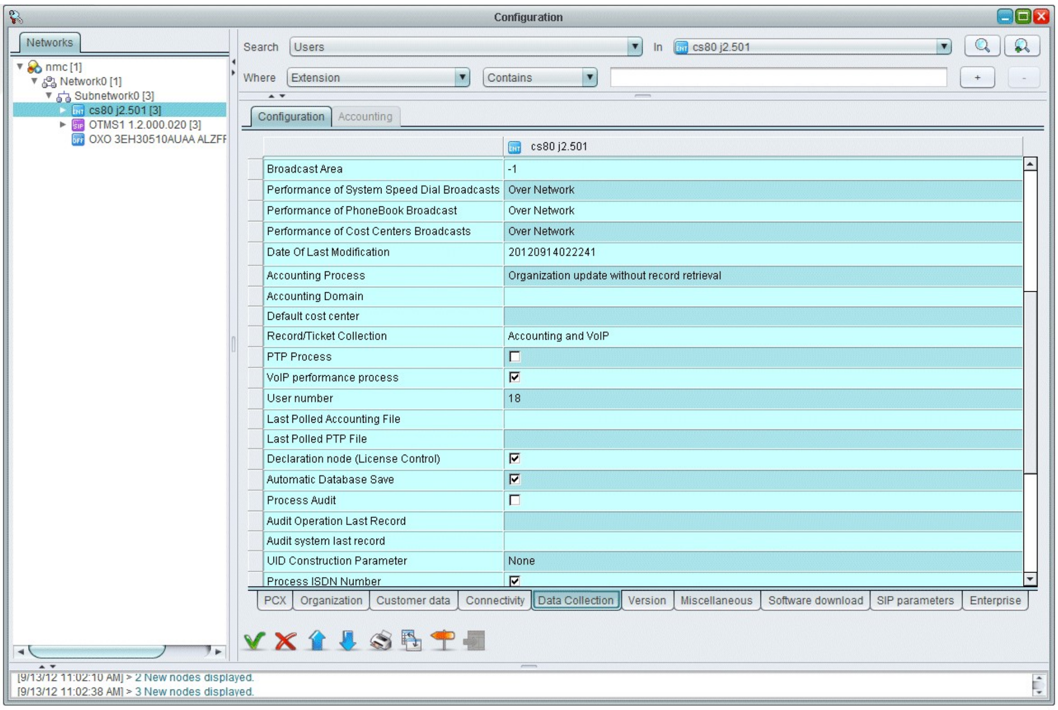
2.4 Selecting Communication Servers for ticket collection

This procedure is used to select

- The Communication Servers for which you want to collect the ticket data
- The type of tickets that you want to recover and store

To activate ticket collection, perform the following operations:

1. In the **Configuration** application, select the Communication Server
2. Click the **Data Collection** tab



3. Double-click the **Record/Ticket Collection** attribute field and select one of the following:
    - **Accounting**: to store accounting files only
    - **VoIP**: to store VoIP files only
    - **Accounting and VoIP**: to store accounting and VoIP files

At synchronization, the files are stored in the directory **8770\data\collector**.

The accounting files are named: TAXAxxxx.DAT (for Alcatel-Lucent OmniPCX Enterprise Communication Server), taxaxxxxx.alz (for OXO), taxaxxxxx.ofc (for OmniPCX Office)

The VoIP files are named: IPXXX.DAI (for Alcatel-Lucent OmniPCX Enterprise CS)

2.5 Configuring filter for external accounting

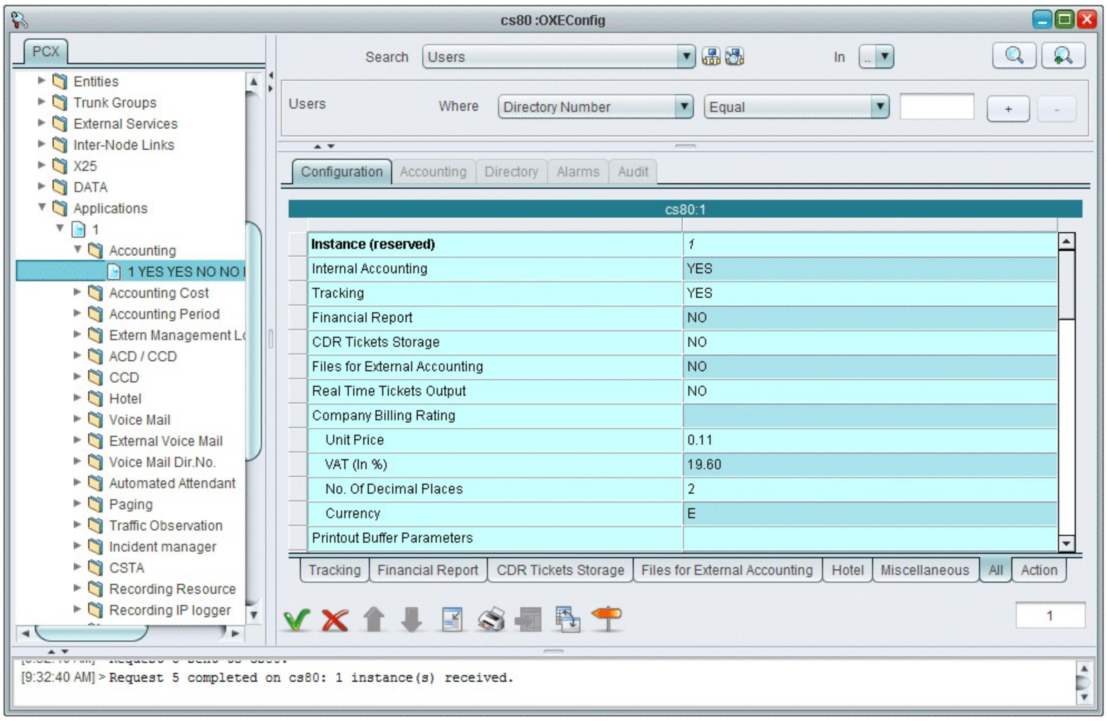
2.5.1 Setting a filter for OmniPCX Enterprise

To add a new filter

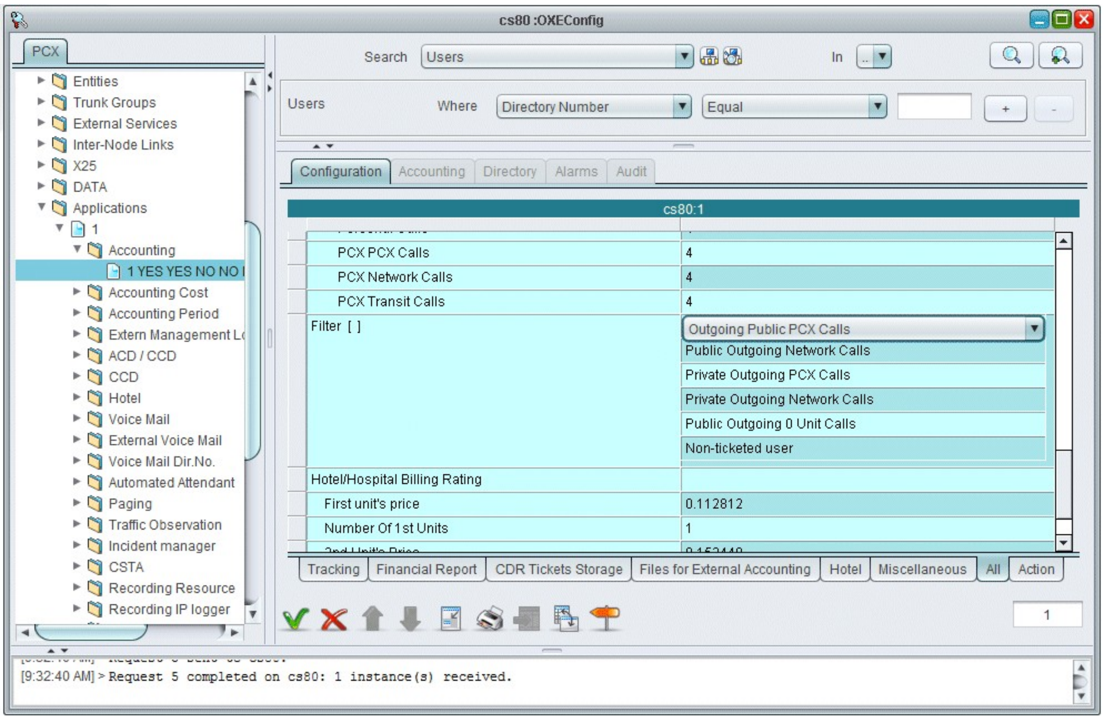
1. In the Configuration application, select the Communication Server, right click: **Configure**

On the object model of the Communication Server:

1. Select the entry **Application** -> **Accounting**
2. Select the **All** tab



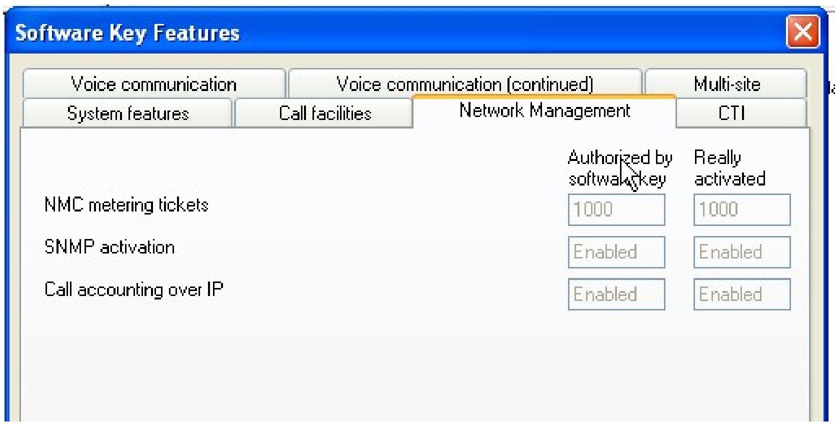
3. Set **Files for external Accounting** field to **Yes**
4. The filter table must be filled with the types of call



2.5.2 Checking license and setting filter for OXO Connect

To check the license for metering tickets:

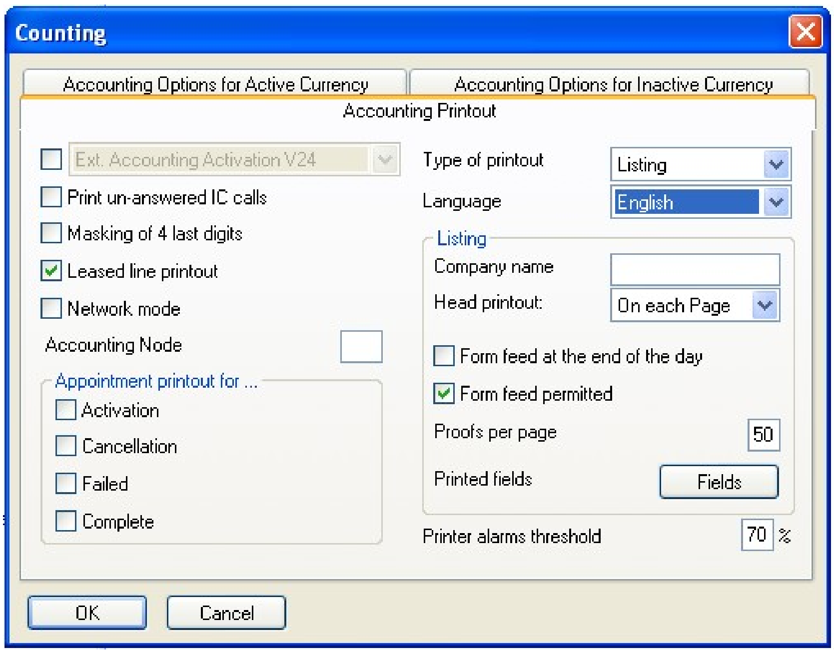
1. In the configuration application, select the Communication Server entry, right click and choose **Configure** /**Online mode** to launch OMC.
2. Select **Hardware and Limits** ->**Software Key Features**
   The Software Key Features window is displayed



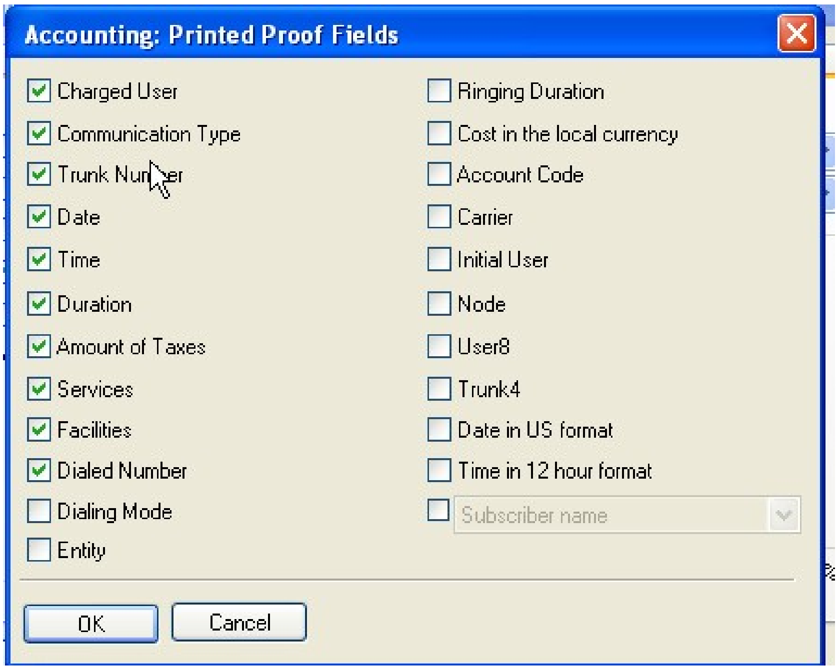
   In the **Network Management** tab, verify that the fields are set to **Enabled** for collecting tickets

To verify or modify the filters:

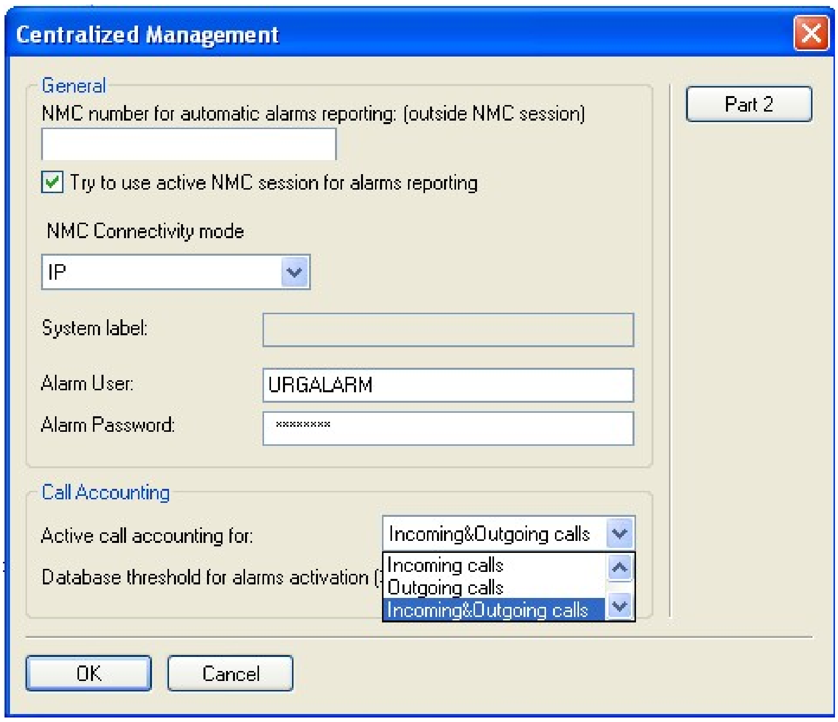
1. From the Communication Server subdirectories, select **Counting** -> **Counting**
   The Counting window is displayed



2. Select the **External Metering Activation** checkbox
3. Validate and click the **Fields** button to display the **Accounting: Printed Proof Fields** window



4. Select all items for all the fields that you want to collect and click **OK**
5. In the main menu, select **Network Management Control** -> **Centralized Management**
   The Centralized Management window is displayed.



6. Select **Incoming&Outgoing calls** for the **Active call accounting**
7. Click **OK**

2.6 Retrieving tickets

There is a schedule task in OmniVista 8770, that launches a synchronization on each Communication Server and retrieves CDR and IP CDR tickets. By default, the task is programmed every night at 1h45.

During synchronization , “Ticket Collector” module retrieves, via ftp, IP*.DAT files (which contain IP tickets data) and TAX*.DAT files (which contain accounting tickets data), and stores these files in the OmniVista 8770 collector folder (this folder could be a local or external repertory set by OmniVista 8770 administrator). By default collector folder path is **\8770\data\collector**.

These files are stored for a defined period in OmniVista 8770 database. This “clean-up” delay (delay in days until the collected data will be deleted) can be changed by theOmniVista 8770 administrator.

These files are presented in an uncompressed form usable by external application. ***.DAT** files are uncompressed (Communication Server proprietary format) and saved on the collector folder as **.DAI** extension under a separate subdirectory named for each Communication Servers as:

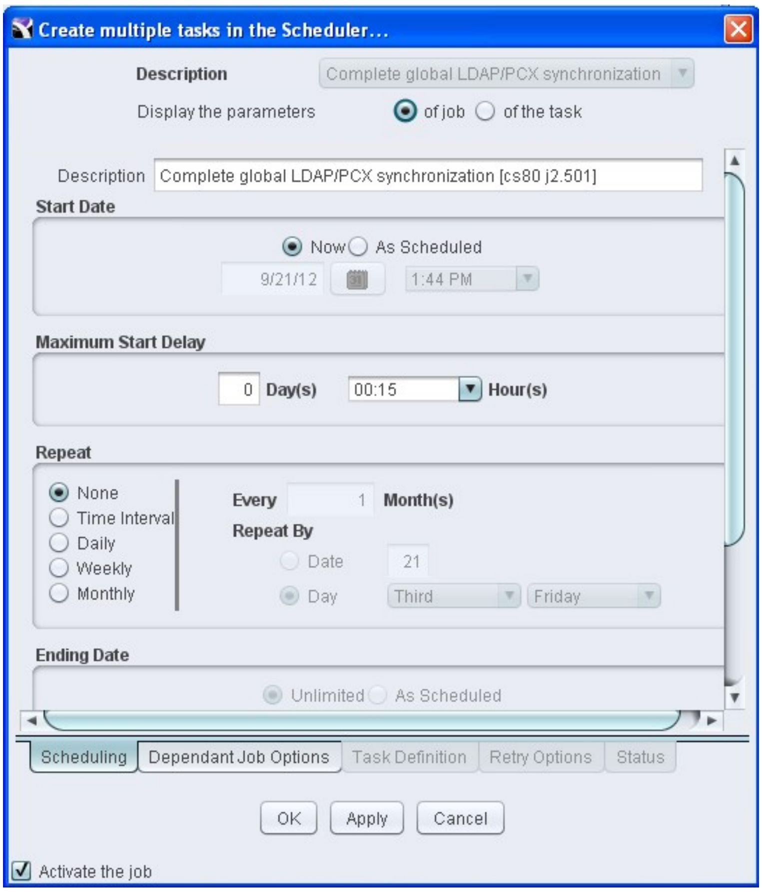
“Networknumber=X\SubnetworknodeNumber=YYY”

2.6.1 Launching a manual synchronization

A manual synchronization can be launched for specific ticket collection.

To launch the manual synchronization:

1. Login as **adminnmc** and launch the **Configuration** application.
2. Select the Communication Server in the networks tree structure.
3. Right click and select **Synchronization** -> **Complete** -> **Global**.
   The **Create multiple tasks in the Scheduler...** dialog box is displayed.



4. Activate the **Now** radio button In the **Start Date** field
5. Click **Apply** and **Close** to launch the synchronization and close the dialog box.

Chapter

3 Ticket formats

3.1 OmniPCX Enterprise Tickets

This section describes the file format for the tickets collected from the OmniPCX Enterprise. How the .DAT file is formatted and an example of a .DAT file

3.1.1 .DAT field format definitions

Each collected file has the same format, so that they can be displayed with a text editor.
There is no column separator, each column has a specific size.
To provide easier visibility, some fields are justified to the right or left

Example: called number start at pos. 6,end pos. 35 Left justified, contains 30 digits maximum.

- Position: position of the 1st and last character of the field in the row
- Justified: position of the text in a line
  - L: left
  - R: right
  - N: not significant

table 3.1: .DAT fields file format for layout display

Field name	position	Size (in charac-ters)	justified
Ticket Version	1,5	5	L
CalledNumber	6,35	30	L
ChargedNumber	36,65	30	L
ChargedUserName	66,85	20	L
ChargedCostCenter	86,95	10	L
ChargedCompany	96,111	16	L
ChargedPartyNode	112,116	5	R
ChargedPartySubaddress	117,136	20	L
CallingNumber	137,166	30	L
CallType	167,168	2	R
CostType	169	1	R
EndDateTime	170,186	17	N
ChargeUnits	187,191	5	R
CostInfo	192,201	10	R
Duration	202,211	10	R
TrunkIdentity	212,216	5	R
TrunkGroupIdentity	217,221	5	R
TrunkNode	222,226	5	R
Personal or Business	227	1	N

Field name	position	Size (in characters)	justified
AccessCode	228,243	16	L
SpecificChargeInfo	244,250	7	N
BearerCapability	251	1	L
HighLevelComp	252,253	2	R
DataVolume	254,263	10	R
UserToUserVolume	264,268	5	R
ExternFacilities	269,308	40	N
InternFacilities	309,348	40	N
CallReference	349,358	10	R
SegmentsRate1	359,368	10	R
SegmentsRate2	369,378	10	R
SegmentsRate3	379,388	10	R
ComType	389	1	N
X25IncomingFlowRate	390,391	2	R
X25OutgoingFlowRate	392,393	2	R
Carrier	394,395	2	R
InitialDialledNumber	396,425	30	L
WaitingDuration	426,430	5	R
EffectiveCallDuration	431,440	10	R
RedirectedCallIndicator	441	1	R
StartDateTime	442,458	17	N
ActingExtensionNumber	459,488	30	L
CalledNumberNode	489,493	5	R
CallingNumberNode	494,498	5	R
InitialDialledNumberNode	499,503	5	R
ActingExtensionNumberNode	504,508	5	R
TransitTrunkGroupIdentity	509,513	5	R
NodeTimeOffset	514,519	5	R
TimeDlt	520,525	6	R
EndOfLine	Unix EOL	0A (10 decimal)	N

table 3.2: Field descriptions for OmniPCX Enterprise

Field name	Description/Value	Use in function of call type			
		Local & network	ABC transit	outgoing	incoming
Ticket Version	ED5.2	X	X	X	X

Field name	Description/Value	Use in function of call type			
		Local & network	ABC transit	outgoing	incoming
Called Number	Depends on the call type	X	-	X	X
Charged Number	Depends on the call type	X	X	X	X
Charged User Name	Depends on the charged number	-	-	X	X
Charged Cost Center	Depends on the charged number	X	-	X	X
Charged Company	Not used	-	-	-	-
Charged PartyNode	100*subnetwork number + node number In heterogeneous networks this field is not filled in if the protocol does not provide it	X	X	X	X
Charged Party Subaddress	Used when ISDN outgoing call otherwise empty	-	-	X	X
Calling Number	Transferring party number after transfer	X	-	X	X

Field name	Description/Value	Use in function of call type			
		Local & network	**ABC transit**	**outgoing**	**incoming**
CallType	0: PublicNetworkOutgoingCall, 1: PublicNetworkOutgoingCall ThroughPrivateNetwork, 2: PrivateNetworkOutgoingCall, 3: LocalNetworkCall, 4: PublicNetworkIncomingCall, 5: PublicNetworkIncomingCall ThroughPrivateNetwork, 6: UnspecifiedCall, 7: PrivateNetworkOutgoingCall ToPublicNetwork, 8: PrivateNetworkOutgoingCall ToPrivateNetwork, 9: PublicNetworkIncomingCall ToPrivateNetwork, 10: PrivateNetworkIncomingCall ToPrivateNetwork, 11:PublicOrPrivateNetworkOutgoingCall ThroughPrivateNet. 12:PublicOrPrivateNetworkIncomingCall ThroughPrivateNet. 13: PrivateNetworkIncomingCall, 14: LocalLocalCall, 15: LocalTransitCall	X	X	X	X
Cost Type	0: Unspecified – ABC-F trunkCall, 1: AnalogTrunkCall, 2: ISDNCircuitSwitchedCall, 3: ISDNPacketCallX25-Bchannel, 4: ISDNPacketCallX25-Dchannel, 5: X25, 6: DigitlNonISDN	-	-	X	-
End Date Time	End communication date or communication transfer Yyyymmdd hh:mm:ss	X	X	X	X
ChargeUnits	Meter pulses received from the public network	-	-	-	X
CostInfo		-	-	X	X
Duration		X	X	X	X
Trunk Identity	Trunk Identification Number (public trunk or private tie-line)	-	-	X	X

Field name	Description/Value	Use in function of call type			
		Local & network	ABC transit	outgoing	incoming
Trunk Group Identity	Trunk Group Number	-	X	X	X
TrunkNode	100*subnetwork number + node number	-	X	X	X
Personal or Business	0: Personal, 1: Business, 2: Normal, 3: Guest	-	-	X	-
Access Code	Project Account Code, Personal Identifier Number or empty	-	-	X	-
Specific ChargeInfo	0: SIOPriorityTrunkGroup, 2: PBXGeneratedChargeUnits, 3: AnalogWithoutChargeUnits, 4: Transcom, 5: AccurateDuration 6: Transgroup	-	-	X	-
Bearer Capability	0: Unspecified, 1: Unrestricted, 2: Speech, 3: Audio3, 4: Audio7, 5: Audio15, 6: Video	-	-	X	-
High Level Comp	0: Unspecified, 1: Telephony, 2: Fax Group3, 3: Mixed Mode, 4: Fax Group4, 5: Teletex Char, 6: Videotex, 7: Telex, 8: MHS, 9: OSI application, 10: Sound, 11: SoundVideotex, 12: SlowScanVideo 13: Not Standard Application	-	-	X	-
Data Volume	Number of transmitted bytes when ISDN communication	-	-	-	-

Field name	Description/Value	Use in function of call type			
		Local & network	ABC transit	outgoing	incoming
User To User Volume	Number of transmitted bytes or number of mini-messages (system option)	-	-	X	-
External Facilities	0: CallingLineIdentificationPresentation, 1: ConnectedLineIdentification Presentation, 2: CallingLineIdentificationRestriction, 3: ConnectedLineIdentification Restriction, 4: MaliciousCallIdentification, 5: CallForwardingUnconditional 6: CallForwardingOnBusy 7: CallForwardingOnNoReply 8: Transfer 9: AdviceOfChargeAtSetup, 10: AdviceOfChargeDuringCall, 11: AdviceOfChargeAtEnd 12: ClosedUserGroup 13: CallWaiting, 14: UserToUserSignalling, 15: UserUserFacility, 21: Mini-messaging, 22: SubAddressing	-	-	X	-
Internal Facilities	5: CallForwardingUnconditional, 6: CallForwardingOnBusy, 7: CallForwardingOnNoReply, 8: Transfer, 23: BasicCall, 24: OperatorFacility, 25: Substitution, 26: PriorityIncomingCall, 27: Transit, 28: PrivateOverflowToPublic, 29: ReroutingPublicToPrivate, 30: FaxServer, 31: VoiceMail, 32: CentralAbbreviatedNumberring, 34: IntegratedServiceVPN, 35: OverflowVPN, 36: ARSService, 37: Disa	-	-	X	-

Field name	Description/Value	Use in function of call type			
		Local & net-work	**ABC transit**	**outgo-ing**	**incom-ing**
Call Refer-ence	Not used	-	-	-	-
Segments Rate1	Not used	-	-	-	-
Segments Rate2	Not used	-	-	-	-
Segments Rate3	Not used	-	-	-	-
Com Type	0 : no specified 1: Voice 2: Data 3: FacilityActivation, 4: FacilityDeactivation	-	-	X	-
X25Incoming FlowRate	0: Unspecified 1: 75 2: 150 3: 300 4: 600 5: 1200 6: 2400 7: 4800 8: 6900 9: 19 200 10: 48 000 11 :64 000 12: > 64 000	-	-	-	-
X25Outgoing FlowRate	0: Unspecified 1: 75 2: 150 3: 300 4: 600 5: 1200 6: 2400 7: 4800 8: 6900 9: 19 200 10: 48 000 11 :64 000 12: > 64 000	-	-	-	-

Field name	Description/Value	Use in function of call type			
		Local & network	ABC transit	outgoing	incoming
Carrier	0: no carrier or unspecified, ARS table Number (1 to 10)	-	-	X	-
InitialDialled Number	Initial dialled number for incoming call or called number when ARS service is used	X	X	X	X
Waiting Duration	With incoming call : duration before answer (in seconds)	-	-	-	X
Effective CallDuration	Only with incoming call	-	-	-	X
Redirected CallIndicator	Only with incoming and ABC network calls : the set who answers isn't the initial dialed number.	X	-	-	X
StartDate Time	Yyyymmdd hh:mm:ss	-	-	X	X
Acting Extension Number	Number of physical charged set. Can be a local number (substitution case), public number (Disa), trunk group number (FSxxx, FPxxx, FCxxx) or node number (SNxxx)	X	-	X	-
Called Number Node	100*subnetwork number + node number	X	-	-	-
Calling Number Node	100*subnetwork number + node number	X	-	-	-
Initial Dialled Number Node	100*subnetwork number + node number	X	X	-	-
Acting Extension Number Node	100*subnetwork number + node number Value 99999 if unknown node; Value XXX99 if only subnet is known	X	-	X	-
Transit Trunk Group Identity	Incoming trunk group number on the transit node	-	X	-	-
Node Tim eOffset	Time difference (in seconds) between 2 nodes; used when the additional CDR is generated on the node of the subscriber; No significant on trunk group node.	-	-	X	X
TimeDlt	Difference between the trunk group time zone ans the system time zone	X	X	X	X
EndOfLine					

3.1.2 Ticket Example

The following example of a TAXAXXXX.DAT was obtained using the `accview –tf` command.

It returns the details of one external outgoing call:

- ext 300 called 305 through an ISDN E1 loop

```
====[/DHS3dyn/account/TAXATUCZ.DAT : Ticket number 1/1/1]=======================
(00) TicketVersion = ED5.2                  (01) CalledNumber = 305
(02) ChargedNumber = 300                    (03) ChargedUserName = IP_300 IP touch
(04) ChargedCostCenter =                    (05) ChargedCompany =
(06) ChargedPartyNode = 3                   (07) Subaddress =
(08) CallingNumber =
(09) CallType = PublicNetworkOutgoingCall
(10) CostType = ISDNCircuitSwitchedCall     (11) EndDateTime = 20090825 09:46:43
(12) ChargeUnits = 0                        (13) CostInfo = 0
(14) Duration = 6                           (15) TrunkIdentity = 1
(16) TrunkGroupIdentity = 0                 (17) TrunkNode = 3
(18) PersonalOrBusiness = Normal            (19) AccessCode =
(20) SpecificChargeInfo =                   (21) BearerCapability = Speech
(22) HighLevelComp = Telephony              (23) DataVolume = 0
(24) UserToUserVolume = 0                   (25) ExternFacilities = CallingLineIdenti-
                                            ficationPresentation
(26) InternFacilities = BasicCall           (27) CallReference = 2
(28) SegmentsRate1 = 0                      (29) SegmentsRate2 = 0
(30) SegmentsRate3 = 0                      (31) ComType = Voice
(32) X25IncomingFlowRate = Unspecified      (33) X25OutgoingFlowRate = Unspecified
(34) Carrier = 0                            (35) InitialDialledNumber =
(36) WaitingDuration = 0                    (37) EffectiveCallDuration = 0
(38) RedirectedCallIndicator = 0            (39) StartDateTime = 20090825 09:46:37
(40) ActingExtensionNumber = 300            (41) CalledNumberNode = 9999
(42) CallingNumberNode = 9999               (43) InitialDialledNumberNode = 9999
(44) ActingExtensionNumberNode = 3          (45) TransitTrunkGroupIdentity = 32767
(46) NodeTimeOffset = 0                     (47) TimeDlt = 0
```

3.2 OXO Connect Tickets

3.2.1 File format

The format is different from the Real time ticket, the Collector uses the native ticket.

This file can be displayed with a Hexa editor. There is a header line (in ASCII mode) with the version of the OmniPCXOXO,Ticketsize, TicketNumber, Network, SubNetwork, NodeNumber.

- PabxRelease=3EH30300FJAA ALZFR200/032.013
- #TicketSize=122
- #TicketNumber=4 (number of tickets in the file)
- #NetworkNumber=0
- #SubNetworkNumber=0
- #NodeNumber=1
- #DataBegin" 0a"

3.2.2 Ticket field definitions

Note that the field with the character + (ie: User Type) is used for only Stand Alone And the character * (ie: Node number) is used only for Network. Networking Mode is defined when the field Accounting Ticket Mode is equal to 03

Field	**byte**	**taxa00018.alz**	**info**
user type	1	00	subscriber Decimal : - **0**: the user is a subscriber - **1**: analog external interface (analog trunk, analog tie line) - **2**: basic public access (T0) or private (dtl0) - **3**: primary public access (t2) or private (dtl2) - **4**: group of subscribers or attendants - **5**: T1 access - **6**: IP trunk Depending on « User type », « ChargedNumber » will be préfixed : - **1**: => prefixe = '**L**' - **2**: => préfixe = '**N**' - **3** and **5** :=> préfixe = '**P**' - **6**: => préfixe = '**V**'
user id	2	20	1 to 8 digits, coded on 8 bytes - USER TYPE = 0 : '0'..'9','A'..'D','*','#' - USER TYPE = 1 : '0'..'9' (<=35) - USER TYPE = 2 : '0'..'9' (<=17) - USER TYPE = 3 : '0'..'7' (<=7) - USER TYPE = 4 : '0'..'9','A'..'D','*','#' - USER TYPE = 5 : '0'..'7' (<=7) - USER TYPE = 6 : '0'..'9' (<=35)
	3	20	
	4	20	
	5	20	
	6	20	
	7	32	2
	8	32	2
	9	36	6

Field	byte	taxa00018.alz	info
call type	10	00	Decimal: - 0 : outgoing call - 1 : incoming call - 2 : transfer of an outgoing call - 3 : transfer of an incoming call - 4 : outgoing call in transit - 5 : incoming call in transit - 6 : connection of a trunk and tie line futher to a transfer of an incoming call - 7 : connection of a trunk and tie line futher to a transfer of an outgoing call - 8 : request of facility - 9 : facility cancel - 10 : outgoing data - 11 : incoming data - 12 : outgoing joining - 13 : outgoing private - 14 : incoming private - 15 : break out 16 : break in - 255 : not defined
trunk line type	11	03	Decimal: - **1**: analog external interface (analog trunk, analog tie line) - **2**: basic public access(t2) private(dtl2) - **3**: primary public access(t2) private(dtl2) - **4**: T1 access - **5**: IP trunk
trunk line ident.	12	30	TRUNK_LINE TYPE = 1 : ‘0’..’9’ (<=35) TRUNK_LINE TYPE = 2 : ‘0’..’9’ (<=17) TRUNK_LINE TYPE = 3 : ‘0’..’7’ (<=7) TRUNK_LINE TYPE = 4 : ‘0’..’7’ (<=7) TRUNK_LINE TYPE = 5 : ‘0’..’9’ (<=35)
	13	20	
	14	30	
day	15	15	String (Date format) : Byte 1 : 1 <= DAY <= 31
month	16	01	Byte 2 : 1 <= MONTH <= 12
year	17	04	Byte 3 : 0 <= YEAR <= 99
second	18	18	Byte 4 : 0 <= SECONDS <= 59
minute	19	34	Byte 5 : 0 <= MINUTES <= 59
hour	20	0C	Byte 6 : 0 <= HOURS <= 23
hour	21	00	String (Hour format) : Byte 1 : 0 <= DUR_HOURS <= 23
minute	22	00	Byte 2 : 0 <= DUR_MINUTES <= 59
second	23	0D	Byte 3 : 0 <= DUR_SECONDES <= 59
pulses (Taxes)	24	00	0…65535

Field	byte	taxa00018.alz	info
	25	01	
service	26	00	, 1, 255 **0**: CCBNT SERVICES (voice service, telefax group **3**, teletex, videotex) **1**: CCBT SERVICES (telefax group 4, data transmission transparent) 255 : not defined
comp. service	27	01	Binary : **Bit 8**: not used **Bit 7**: disa transit **Bit 6**: transfert **Bit 5**: pbx forwarding **Bit 4**: not used **Bit 3**: on line metering **Bit 2**: external diversion **Bit 1**: user to user signaling
dialled digits	28	0A	**Byte 1**: number of digits (0 to 26) **Byte 2 to byte 14**: digits 1 octet codified on 2 digits.
	29	00	
	30	00	
	31	00	
	32	00	
	33	00	
	34	00	
	35	00	
	36	00	
	37	02	02
	38	98	98
	39	14	14
	40	39	39
	41	08	08
dial.mode	42	00	Decimal : **0**: manual dialling **1**: individual short code **2**: short code mode **255**: not defined Dialling mode
ring time	43	00	Decimal : **Byte 1**: 0 <= SECONDS <= 59 **Byte 2**: 0 <= MINUTES <= 59 Ringing time for incoming calls.

Field	byte	taxa00018.alz	info
	44	00	
cost	45	00	**Byte 1** and **byte 2** : integer part **Byte 3** and **byte 4** : factional part (in 1/10000) Cost is given in local currency
	46	00	
	47	00	
	48	00	
account-code	49	00	**Byte 1**: number of digits (0 to 16)
	50	00	**Byte 2** to b**yte 14**: digits Same coding as « DIALLED DIGITS » If account code is filled in the field « PersonalOrBusiness » is set to 1 (Business).
	51	00	
	52	00	
	53	00	
	54	00	
	55	00	
	56	00	
	57	00	
subscriber name	58	20	**Byte 1** to **byte 16** Calling party name (for outgoing call) or Called party name (for incoming call) The first name or the name are separated by a blank.
	59	20	
	60	20	
	61	70	P
	62	6F	o
	63	73	s
	64	74	t
	65	65	e
	66	32	2
	67	32	2
	68	26	6
	69	4F	O
	70	58	X
	71	4F	O
	72	52	R
	73	32	2

Field	byte	taxa00018.alz	info
init.user type	74	FF	Decimal : **0**: the user is a subscriber **1**: analog external interface (analog trunk, analog tie line) **2**: basic public access (T0) or private (dtl0) **3**: primary public access (t2) or private (dtl2) **4**: group of subscribers or attendants **5**: T1 access **6**: IP trunk
init. User ident.	75	20	1 to 8 digists, coded on 8 bytes USER TYPE = 0 : ‘0’..’9’,‘A’..’D’,’*’,’#’ USER TYPE = 1 : ‘0’..’9’ (<=35) USER TYPE = 2 : ‘0’..’9’ (<=17) USER TYPE = 3 : ‘0’..’7’ (<=7) USER TYPE = 4 : ‘0’..’9’,‘A’..’D’,’*’,’#’ USER TYPE = 5 : ‘0’..’7’ (<=7) USER TYPE = 6 : ‘0’..’9’ (<=35) This info element contains in case of : - **outgoing call** a blank field or after transfer the same information as incoming calls - **incoming call** the number of subscriber or group initially called - **break-out** the remote calling charged number.
	76	20	
	77	20	
	78	20	
	79	20	
	80	20	
	81	20	
	82	20	

Field	byte	taxa00018.alz	info
init. User ident. (in Network)	83	20	Byte 1 : number of digits Byte 2 to byte 14 : digits 1 octet code on 2 digits. This info element contains in case of : **outgoing call** the number gived by the first numbering plan translation from the number initially dialled by the user or after transfer it has the same signifiaction as incoming call, **incoming call** the private or public number of subscriber or group initially called, **break-out** the remote calling charged number. The user identification in networking mode can be up 1 to 26 digits : it is the « networking address » of the caller or called depending of the direction of the call. **Byte 1**: number of digits **Byte 2 to byte 14**: digits 1 octet code on 2 digits. Filled an padded with spaces only in networking mode. This info element contains in case of : **outgoing call** the caller number, **incoming call** the private or public number of subscriber which has answered, break-out the remote calling charged number. The user identification in networking mode can be up 1 to 26 digits : it is the « networking address » of the caller or called depending of the direction of the call
	84	20	
	85	20	
	86	20	
	87	20	
	88	20	
	89	20	
	90	20	
	91	20	
	92	20	
	93	20	
	94	20	
	95	20	
	96	20	

Field	byte	taxa00018.alz	info
User Ident (in Network)	97	20	number of digits
	98	20	
	99	20	
	100	20	
	101	20	
	102	20	
	103	20	
	104	20	
	105	20	
	106	20	
	107	20	
	108	20	
	109	20	
	110	20	
Net. Comp.Ser.	111	00	Binary : **Bit 8**: not used **Bit 7**: not used **Bit 6**: not used **Bit 5**: Forced on net **Bit 4**: Overflow **Bit 3**: ARS/LCR **Bit 2**: VPN(virtual private network) **Bit 1**: user to user signaling (ISVPN)
node number	112	20	0…127], 255 is reserved to the **unanswered incoming call**.
carrier	113	20	Character (ASCII value)
acc. ticket mode	114	01	Decimal : 1 : WRITTEN PROOF 2 : TICKET 3 : NETWORK
sent UUI counter	115	00	
	116	00	
	117	00	
	118	01	
rec. UUI counter	119	00	
	120	00	
	121	00	
	122	00	

3.2.3 Ticket example

The folllowing example of the native file taxa00018.alz shows:

- an outgoing call from the 226
- an incoming call to the 221


```
23 50 61 62 78 52 65 6c 65 61 73 65 3d 33 45 48 33 30 33 30 30 46 4a 41 41 20 41 4c 5a
46 52 32 30 30 2f 30 33 32 2e 30 31 33 20 20 20 20 20 20 20 20 20 20 20 20 0a 23 54 69 63
6B 65 74 53 69 7a 65 3d 31 32 32 20 20 20 20 20 20 20 0a 23 54 69 63 6b 65 74 4e 75 6D
62 65 72 3d 32 20 20 20 20 20 20 20 20 20 20 0a 23 4e 65 74 77 6f 72 6b 4e 75 6d 62 65 72
3d 30 20 20 20 20 20 20 20 20 20 20 0a 23 53 75 62 4e 65 74 77 6f 72 6b 4e 75 6d 62 65 72
3d 30 2à 20 20 20 20 20 20 20 20 0a 23 4e 6f 64 65 4e 75 6d 62 65 72 3d 31 20 20 20 20
20 20 20 20 20 0a 23 44 61 74 61 42 65 67 69 6e 0a 00 20 20 20 20 20 32 32 36 00 02 30
20 30 15 01 04 18 34 0c 00 00 0d 00 01 00 01 0a 00 00 00 00 00 00 00 00 02 98 14 39 08
00 00 00 00 00 13 88 00 00 00 00 00 00 00 00 00 20 20 20 70 6f 73 74 65 32 32 36 4f 58
4f 52 32 ff 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20
20 20 20 20 20 20 20 20 20 20 20 00 20 20 01 00 00 00 01 00 00 00 00 00 20 20 20 20 20
32 32 21 01 03 30 20 30 15 01 04 28 3b 0C 00 00 06 00 00 00 00 00 00 00 00 00 00 00 00
00 00 00 00 00 00 00 06 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 20 70 6f 73 74 65 32
32 31 49 50 4f 58 4f 52 32 00 20 20 20 20 20 32 32 31 20 20 20 20 20 20 20 20 20 20 20
20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 00 20 20 01 00 00 00 00 00 00 00 00
```

```
#PabxRelease=3EH30300FJAAALZ
FR200/032.013            #Tic
ketSize=122        .#TicketNum
ber=2          .#NetworkNumber
=0          .#SubNetworkNumber
=0          .#NodeNumber=1
.#DataBegin..      226..0
0.....4 ..................................9.
..............................   poste226OX
OR2y...........................................
.
221..0 0...(;...................................
......................................poste2
21IPOXOR2.    221
.   .........
```


3.3 IP tickets format for OmniPCX Enterprise

The format for the IP ticket is different from the Real time ticket, the Collector uses the native ticket. The Communication Server VoIP Statistics Tickets data file uses Type-Length-Value (TLV) format (Type=1 Byte, Length=2 Byte, Value). This file can be displayed with a Hexa editor.

Information about the IP tickets and their applicability is provided in table: IP Ticket parameter IDs, Definitions and the Concerned Equipment

Note:
The last column shows fields that can be used by a 3rd party application (Alcatel-Lucent Partner). By using the TLV mechanism the IP equipment can, with some exceptions, decide to use a parameter or not, either because it is not implemented or not significant.

3.3.1 IP Ticket Fields

The following table contains the data field definitions for OmniPCX Enterprise equipment. Section IP Ticket Fields Definitions explains the ID numbers and how the information can be used.

table 3.5: IP Ticket parameter IDs, Definitions and the Concerned Equipment

			Concerned equipment							
ID	**TLVID (hexa)**	**Length (Byte)**	**GA/ GD**	**INTIP**	**IP-Phone V2**	**IPTouch**	**Soft-phone**	**eVA**	**Range Limit value (decimal**	**3rd Party use**
1	01	4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
2	02	2	**X**	**X**	**X**	**X**	**X**	**X**	**0.99**	**X**

			Concerned equipment							
ID	**TLVID (hexa)**	**Length (Byte)**	**GA/ GD**	**INTIP**	**IP-Phone V2**	**IPTouch**	**Soft-phone**	**eVA**	**Range Limit value (decimal**	**3rd Party use**
3	03	1	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
4	04	1	**X**	**X**				**X**	**0..128**	**X**
5	05	1	**X**	**X**				**X**	**0..52**	**X**
6	06	2	**X**	**X**	**X**	**X**	**X**	**X**	**1..4[0..2]**	**X**
7	07	1	**X**	**X**				**X**	**/**	**X**
8	08	4 to 16	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
9	09	4 to 16	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
10	0A	30	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
11	0B	30	**X**	**X**	**X**	**X**		**X**	**/**	**X**
12	0C	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
13	0D	4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
14	0E	4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
15	0F	1	**X**	**X**	**X**	**X**	**X**	**X**	**0..3**	**X**
16	10	1	**X**	**X**			**X**		**0..1**	
17	11	1	**X**	**X**			**X**		**0..1**	**X**
18	12	1			**X**	**X**	**X**		**80..84**	**X**
19	13	1	**X**	**X**	**X**	**X**	**X**	**X**	**20, 30, 40**	**X**
20	14	1	**X**	**X**	**X**	**X**	**X**	**X**	**20, 30, 40**	**X**
21	15	1	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
22	16	4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
23	17	4	**X**	**X**	**X**	**X**	**X**		**/**	**X**
24	18	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
25	19	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	
26	1A	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
27	1B	5*2	**X**	**X**	**X**	**X**	**X**	**X**	**not 65535**	**X**
28	1C	5*2	**X**	**X**	**X**	**X**	**X**	**X**	**not 65535**	**X**
29	1D	1	**X**	**X**	**X**	**X**	**X**	**X**	**/**	
30	1E	10*2	**X**	**X**		**X**			**/**	**X**
31	1F	5*2	**X**	**X**	**X**	**X**			**/**	**X**
32	20	10*4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
33	21	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	

			Concerned equipment							
ID	**TLVID (hexa)**	**Length (Byte)**	**GA/ GD**	**INTIP**	**IP-Phone V2**	**IPTouch**	**Soft-phone**	**eVA**	**Range Limit value (decimal**	**3rd Party use**
34	22	4	**X**	**X**					**/**	
35	23	4*2	**X**	**X**					**/**	
36	24	2	**X**	**X**					**/**	
37	25	10*4	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
38	26	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	
39	27	X			**X**	**X**		**X**	**/**	
40	28	2	**X**	**X**	**X**	**X**	**X**	**X**	**0..255**	**X**
41	29	1	**X**	**X**	**X**	**X**	**X**		**10,20,30**	**X**
42	2A	2	**X**	**X**	**X**	**X**			**/**	**X**
43	2B	1	**X**	**X**					**0..1**	
44	2C	1	**X**	**X**					**0..1**	
45	2D	2	**X**	**X**	**X**	**X**	**X**	**X**	**/**	**X**
46	2E	1	**X**	**X**	**X**	**X**	**X**	**X**	**0..1**	
47	2F	1	**X**	**X**	**X**	**X**	**X**	**X**	**0..7**	
48	30	2	**X**	**X**	**X**	**X**	**X**	**X**	**0..4094**	
49	31	1	**X**	**X**	**X**	**X**	**X**	**X**	**0..63**	
55	37	1	**X**	**X**	**X**	**X**	**X**	**X**	**0..1**	
60	3C	1	**X**	**X**	**X**	**X**	**X**	**X**	**/**	
61	3D	5*2				**X**			**/**	**X**
62	3E	5*2				**X**			**/**	**X**
80	50	1							**0..1**	
81	51	1							**/**	
82	52	1							**/**	
83	53	1							**0.1**	
84	54	1							**/**	
85	55	4							**/**	
86	56	4							**/**	
87	57	4							**/**	
89	59	2							**/**	
90	5A	2							**/**	

3.3.2 IP Ticket Fields Definitions

In the following sections:

- Fields are numbered according to their IDs

- They are regrouped into different sections according to the type of information that can be collected

3.3.2.1 *Localization of the call on the time:*

Field [1]: End of communication

Date of end of communication corresponds to the number of seconds (in linux format) from 1970 at 00:00:00GMT to which we add the difference in time (according to the meridian and taking into account summer/winter times).

This field is filled by the CPU on reception of the ticket.

The hour on the ticket is the local hour of the Communication Server which has produced the ticket. It is mandatory to synchronize all the Communication Server network with the same hour. If the Communication Servers involved on the call are not configured on the same Time Zone, the difference between different time zones must be take into account to deduce tickets for the same communication.

This field could be used to retrieve all tickets for the same communication

The start time can be derived subtracting the call duration using the “Duration” field (field [12]):

```
“Date of beginning of communication” = “date of end of communication”
– “duration of communication”
= “field [1]” – “field [12]”
```

3.3.2.2 *Localization of end-points of the segment:*

Field [2]: Node Number

Node Number on which the IP equipment (coupler/IP-Phone) is found

This field is filled by the CPU during reception of the message.

Field [3]: Protocol Version

Protocol version used between the coupler and the CPU for the ticket

Actual values: Up to release 6 : 0 (there are equipments which use 1) From release 7 : 2

Field [4]: Crystal Number

Rack number on which the coupler is found

If the coupler is an eVA, the value will always be 18 (0x12)

Field [5]: Coupler Number

Coupler (IP Media gateway device) Number

It the coupler is an eVA, the value will be always 0

Field [6]: Equipment type

Type of equipment from which ticket is generated

Format:

- `Byte 1: Type:1=IPP|2=APC|3=CplOmEnt|4=CplOmOFF] :`
- `Byte 2: [Type:1=IPPv2|2=NOEIPP|0=4980|1=WSftIP|0=IntIP|1=GD|2=eVA] :`

  `Byte 1` indicates type of equipment

  - **1=IPP**: corresponds to an IP Phone

- **2=APC**: corresponds to a PC application
- **3=CplOmEn**t: corresponds to an OmniPCX Enterprise coupler
- **4=CplOmOFF**: corresponds to an OXO Connect coupler.

`Byte 2` gives complementery details about the type of equipment indicated in the first byte:

- For IP phones (equipment type 1):
  - **1=IPPv2**: e-Reflexes (IP phone v2)
  - **2=NOEIPP**: IP Touch
- For PC applications (equipment type 2):
  - **0=4980**: 4980 Softphone (PCMM2)
  - **1=WSftIP**: WebSoftPhone IP
- For OmniPCX Enterprise couplers (equipment type 3):
  - **0=IntIP**: INTIPA or INTIPB board
  - **1=GD**: GD or GA board
  - **2=eVA**: 4645 voice mail

Field [7]: Timeslot Number

Time slot (channel) number used to help finding the DSP used by simple deduction.

3.3.2.3 *Identification of end-points of the segment:*

Field [8]: Local IP address

This field contains the IP source address of the IP equipment transmitting the IP flow. This equipment can be an IP Phone, an INTIP board, a GD/GA board ... (refer to equipment type possibilities on field [6]). For each RTP flow, a fast socket is created. The content includes the addresses of IP source and IP destination.

Field [9]: Distant IP address

This field contains the IP address of the destination (receiving) IP equipment of the IP flow. This equipment can be an IP Phone, INT-IP board, a GD/GA board ... (refer to equipment type possibilities on Field [6]).

3.3.2.4 *Identification of calling/called set:*

Field [10]: Local ID

This field is not always completed. When this is possible, it contains the directory number of the local set involved in the call (information sent by CPU).

Field [11]: Distant ID

This field is not always completed. When this is possible, it contains the directory number of the remote (distant) set involved in the call (information sent by CPU).

Specificity:

- Tickets generated for ringback tone could be identified by comparing local ID and Distant ID on the 2 symmetric tickets: if local ID and distant ID are the same on the 2 symmetric tickets (tickets from the same segment) we can deduce that this is a segment used for ringback tone
- For tickets generated for ringback tone we can know which phone have initiated the call. In

fact, on tickets generated on both sides the “Local ID” field will be equal to calling party MCDU and the “Distant ID” field will be equal to called party MCDU

3.3.2.5 Communication duration of the IP segment (RTP flow):

Field [12]: Call Duration

This field shows call duration in seconds during which the equipment has sent (and received) packets on the IP

Network (IP segment). The time elapsed during the ringing phase may be included in this time (refer to “user case” annex)

For INTIP/GD call duration value is determined as following: 0-999msec = 0sec, 1000-1999msec = 1sec. ...

3.3.2.6 Identification of the segment:

Field [13]: Local SSRC

This field is the value of the “local synchronisation source identifier “ that allows to identify an IP communication segment (IP segment processed by IP ticket) The SSRC identifier is a randomly chosen value meant to be globally unique within a particular RTP session

Field [14]: Distant SSRC

This field is the value of the “distant synchronisation source identifier “ that allows to identify an IP communication segment (IP segment processed by IP ticket) The SSRC identifier is a randomly chosen value meant to be globally unique within a particular RTP session

3.3.2.7 VoIP parameters that could have an impact on voice quality:

Field [15]: Algo Compression Type

`0=G711A|1=G711U|2=G723|3=G729`

This field contains the type of compression algorithm used:

- **0**: G711 PCMA (64 kbps -> A law: Europe)
- **1**: G711 PCMU (56 kbps -> µ law: USA)
- **2**: G723.1 (6.3 kbps)
- **3**: G729 A (8 kbps)

Possible values per equipment:

INTIP/GA/GD/IPP/IPT : 0,1,2,3
eVA : 0,1

This value will depend of codec management

Remark:
At the H323 protocol level, we do not distinguish the cases G723 5.3 kbps and 6.3 kbps. At the mao end we also have a single value for G723 and the CPU indicates G723 6.3kbps to the coupler (H323 gateway).

In the same way on reception of G723 indication by the distant gateway, the coupler indicates G723 6.3 kbps/s to the CPU in all cases. The coupler always initialises DSP with G723 6.3kbps. In the case of connexion to a CISCO gateway configured in 5.3kbps, we note that the frames received by INTIP are made up of 20 octets (5.3kbps) and those sent by INTIP are made up of 24 octets (6.3kbps)

(CDHva60191)

The quality is affected, but remains correct. In this case, we could send up the value 4 instead of 2 To be seen (If we decide to do this, a RA will be created for the concerned cases).

Field [16]: VAD

This field shows if silence suppression (voice activity detection) is enabled

- **false**: disabled,
- **true**: enabled

Voice Activity Detection (VAD) allows the bandwidth used to be reduced. When VAD is requested, the background noise generator is systematically enabled.

VAD is enabled via two system options, one applies to calls coded in G711, the other to calls coded in G723 or G729. In the case of an H.323 call, VAD involves negotiation between the caller and the called party.

This parameter is manageable on OmniPCX Enterprise (path: “System/ Other System Param./ Compression Parameters/ VAD on G711 and Voice Activity Detect (Comp Bds)”)

Field [17]: Echo Canceler

Indicate in which state is the set: handset /Hands-free / loud speaker

Possible values depending on IP phone type (IPP, IPT):

- Idle 0x50
- Handset 0x51
- Group-listening 0x52
- On-hook-dial 0x53
- Hands-free 0x54
- Loudspeaker Announce not applicable - Ringing not applicable

Field [18]: Voice Mode

Indicate in which state is the set : handset /Hands-free / loud speaker

Possible values depending on IP phone type (IPP, IPT):

- Idle 0x50
- Handset 0x51
- Group-listening 0x52
- On-hook-dial 0x53
- Hands-free 0x54
- Loudspeaker Announce not applicable
- Ringing not applicable

3.3.2.8 Parameters on RTP packet flow that could be used to calculate VoIP QoS:

Field [19]: Requested Framing Duration

Framing is the time (in ms) that corresponds to the duration of an RTP voice packet transmitted on the IP network.

For example, 20 ms, 30 ms, 40 ms,… (e.g. for 4645 voice mail, it is always 30 ms). This field shows the real duration in ms of framing transmission to the IP network. This framing may be different to the system configuration following negotiation with the destination equipment.

This information is presented in the ticket in such a manner that we can know which framing sent to the equipments do not generate the statistics ticket IP (H323 call etc...).

Note 1:
There can be a difference between the "requested" framing given by the CPU and the real framing, because a negotiation can be done between the IP equipment. This field will be the framing used

The framing value is selected separately for each algorithm using the following Parameters:

- *G711 VOIP Framing : Possible values:*
  - *20ms (default value)*
  - *30ms*
- *G729 VOIP Framing : Possible values:*
  - *20ms (default value)*
  - *30ms*
  - *40ms*
- *G723 VOIP Framing : Cannot be modified: 30ms*

Field [20]: Received Framing

This field corresponds to the duration of RTP voice packet received expressed in ms.

Note 2:
with VAD, the frames received from the network may have a smaller size in the silent mode.

Example:

G711 and IP framing of 30 ms:

Normal frame in voice mode = 240 bytes (30msec of speaking) In silent mode = 160 bytes or 80 bytes (20ms or 10ms of speaking) with a frame of 1 byte (SID) which follows.

(In the case of INTIP coupler in IP to IP 20ms, we may have the case of 81 bytes = voice + SID in the same frame)

Field [21]: Framing Change NB

Number of times IP framing of communication has changed

Note 3:
As for the field 0x20, take care to see if VAD is activated and if not increment this field every time the received packet changes size.

Field [22]: RTP Received Packets NB

Total number of RTP packets received by the IP equipment for this communication, including SID packets.

Alcatel-Lucent cannot guarantee that this value represents the real traffic exchanged on the IP network.

Field [23]: Total RTP Packets Sent

Total number of RTP packets sent by the IP equipment for this communication including SID packets.

Alcatel-Lucent cannot guarantee that this value represents the real traffic exchanged on the IP network.

Field [24]: RTP Lost Packets NB

Total number of RTP packets lost by the IP equipment on reception for this communication.

It is possible to deduce a QoS element by calculation of the percentage of number of lost RTP packets:

% lost RTP packets= [24] / [22] * 100

If “% lost RTP packets” < 1% => Good level

If 1% >“% lost RTP packets” < 2% => Acceptable level

If 2% >“% lost RTP packets” < 4% => Fair level

If “% lost RTP packets” > 4% => Poor level

If this QoS is poor, an analysis for the RTP packets lost are concentrated on a short time interval and to check the consecutive BFI level

Field [25]: Total Silence

CNG (Comfort Noise Generation) counter:

In the silence suppression mode, the DSP will stop sending voice packets when the voice level goes below a certain level, it will then send 1 SID frame with information on the background noise and no other frame before the voice level becomes higher than the background level.

The SID is used for a “Comfort Noise Generation” on the opposite side. In some cases more than a normal number of SID frames could be sent. For the G729 case, it is a known problem due to the standard. This is not necessarily the number of seconds of no speech, but the time where the DSP has generated a comfort noise. With a really noisy environment, it is possible to have 0 seconds of CNG, even if no person is speaking.

Field [26]: NB SID Received Packets

Number of SID frames received for this communication.

The SID (Silence IDentification) are the frames which contain the information on the background noise . This information is used to generate a background noise at the receivers end.

VAD should be validated in order to have the SID packet.

Field [27]: Delay

```
[0-40]:0 | [40-80]:0 | [80-150]:0 | [150-250]:0 | [+250]:0 |
```

table: Delay Range represents the distribution of the round trip delay (2-way accumulated) on this communication.

table 3.6: Delay Range

Range (ms)	Level
0-40	Very Good
40-80	Good
80-150	Acceptable
150-250	Fair

250 and more	Poor

When you design networks that transport voice over packet, frame, or cell infrastructures, it is important to understand and account for the delay components in the network. If you account correctly for all potential delays, it ensures that overall network performance is acceptable. Overall voice quality is a function of many factors that include the compression algorithm, errors and frame loss, echo cancellation, and delay. The International Telecommunication

The International Telecommunication Union (ITU) considers network delay for voice applications in Recommendation G.114. This recommendation defines three bands of one-way delay as shown in table: ITU Delay Recommendations next table:

table 3.7: ITU Delay Recommendations

Range (ms)	**Description**
0-150	Acceptable for most user applications
150-400	Acceptable provided that administrators are aware of the transmission time and the impact it has on the transmission quality of user applications
Above 400	Unacceptable for general network planning purposes. However, it is recognized that in some exceptional cases this limit is exceeded

Round trip delay is calculated at regular intervals and based on the exchange of RTCP packets.

Note 4:

The information is supplied via RTCP packets and these are normally given every 6s. So a communication having a duration less than 6s will not have received any RTCP packets and the IP equipment will/should therefore not include the [27] field in the ticket.

Delay ranges:

*0-40 : 0*125us - 320*125us (40 ms included)*

*40-80 : 321*125us - 640*125us (80 ms included)*

etc ...

Field [28]: Max Delay

Maximum round trip delay measured in the network during the call for this IP segment.

The measurement is based on the exchange of RTCP packets and has an uncertainty of around 10ms.

If “Max delay” is up to 150 ms that means that there is a bad QoS. Field [27] (Delay) must be analyzed in order to know if RTD was up to 150 ms on many occasions. If this is the case, a network problem could exist.

Note 5:

The information is supplied via RTCP packets and these are normally given every 6s. So a communication having a duration less than 6s will not have received any RTCP packets and the IP equipment will/should therefore not include the [28] field in the ticket.

Field [29]: NB DTMF Detected

DTMF signals may be carried directly in-band, and thus in the RTP flow.

This field corresponds to the number of DTMF packets detected during the communication (for Gateway => number of DTMF and for remote => Number of packets of type DTMF).

Note 6:

The only reason a DTMF number is mentioned is to explain an abnormally high BFI number. In fact the DTMFs generate BFIs.

This field is to be used for other things only as a complement of the BFI number.

3.3.2.9 VoIP parameters that could be used to calculate VoIP QoS:

Field [30]: Consecutive BFI

[1]	: 0
[2]	: 0
[3]	: 0
[4]	: 0
[5]	: 0
[6]	: 0
[7]	: 0
[8]	: 0
[9]	: 0
[10+]	: 0

This table of 10 values represents the occurrences of consecutive BFIs (Bad Frame Interpolation) played. It does not apply to IPTouch sets and 4645 voice mail.

BFIs may be linked with two types of problems:

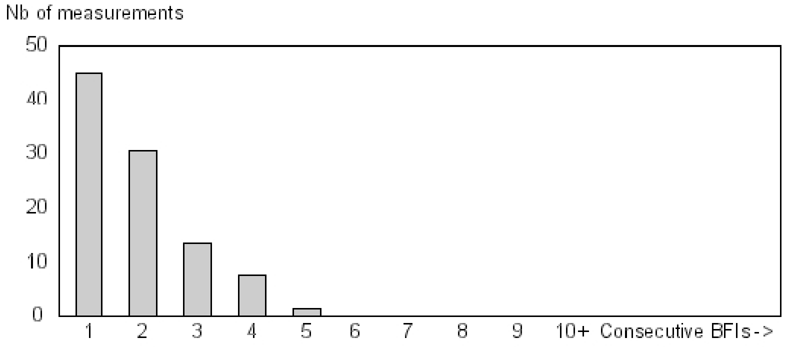
- jitter problems
- IP packet loss problems (hole time) which does not introduce delay

Every BFI indicates that at a given moment the DSP did not receive a packet , and has extrapolated . See the field DSP_FRAMING_DURATION (field [41]) for the unit used.

In general, due to the non-synchronisation between the starting and stopping of IP flow we could have faulty counters.

For starting we must start our counters only on receiving the first IP packet (refer to remark 1), but nothing is done for stopping.

For the consecutive BFIs, we should not have a problem at the end of communication, as they are counted on receiving the first voice packet after BFI , and there we will not have a voice packet



Consecutive BFI level according to codec used			
Range if G711 codec	**Range if G723 codec**	**Range if G729 codec**	**Level**
No BFI	No BFI	No BFI	Good
1 BFI to 3 BFI	1 BFI	1 BFI to 3 BFI	Acceptable
4 BFI to 6 BFI	Not used	Not used	Fair
7 BFI to 10 BFI and more	2 BFI to 10 BFI and more	4 BFI to 10 BFI and more	Poor

Remark:
The different counter should not start before the first RTP packet from the IP network has been received.

Caution:
This could hide a real problem: In the IPPhone there was a problem on a version and which could not be analyzed with the help of QOS. Ticket with this technique The problem was as follows. At the beginning of the communication we did not receive any answer to the first ARP request. In order to know the MAC address of the distant, we had to wait for the answer to our second request This produced 1 second of silence at the beginning of the communication . If the problem re-occurs, the QOS ticket will indicate that the communication is perfect, however the user will not be satisfied.

Field [31]: BFI Distribution

This table of 5 values represent the distribution of the BFIs. It does not apply to IP Touch sets and 4645 voice mail.

BFI and voice play duration are measured in 10s time intervals. The ratio of duration of BFIs played/duration of received voice is expressed in percentage.

Voice play duration is used and not the interval (10s), otherwise Voice Activity Detection (VAD) distorts the result.

`[0%] : 0`	equal to 0%
`[<0-1] : 0`	more than 0% until to 1%
`[<1-2] : 0`	more than 1% until to 2%

[<2-3] : 0	more than 2% until to 3%
[<3%+] : 0	more than 3%

Field [32]: Jitter Depth

[0]:0 | [1]:0 | [2]:0 | [3]:0 | [4]:0 |

[5]:0 | [6]:0 | [7]:0 | [8]:0 | [9]:0 |

This table of 10 values shows the distribution of jitter size. The table is incremented each time a packet is sent to the DSP (at DSP framing frequency).

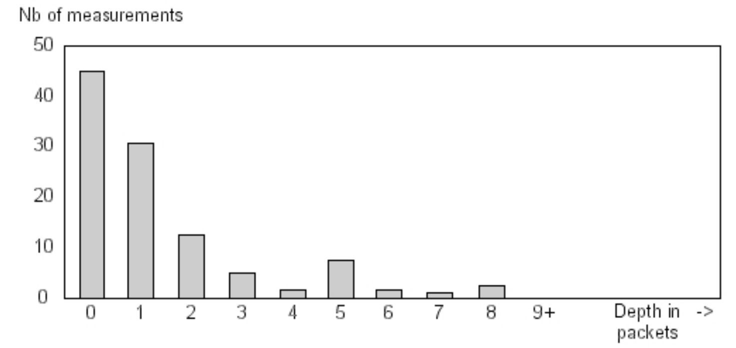
This measurement process depends on kind of IP equipment (local party) involved on this call:

Two different measurement process is used according to equipment involved:

- One for the INTIP/GD/GA with a jitter buffer size static
- Another for the 4980 application and the IP Phone with a jitter buffer size dynamic

It does not apply to 4645 voice mail (eVA).

- The following table shows jitter depth for INTIP/GD/GA
  With a static buffer, the INTIP/GD/GA should continue to use the buffer depth distribution, since otherwise the client could think that he just had one problem of delay at one point in the conversation, and not a problem of delay lasting from the burst till the end of the conversation. See field [37] for the network jitter distribution.



The default size of the jitter buffer for INTIP and GD/GA coupler is 100 ms (It was 200 ms for LIOE). The size changing could be done when the network do not have so good characteristics and the people complain that they lost voice syllables or got tremulous voice. The following parameters could be changed only by OXE administrator:

- For INTIP coupler:

  It could be change and taken into account dynamically as described below by connecting to INTIP board:

  - INTIP:mcc
  - MCC:jitterdepth
  - Max Jitter Sz (ms) :64 <-- Enter an Hexa value
  - MaxVoiceJitterSz 100 <-- the result is 100 ms
  - 0000008C-03D7474F: End of command execution

  If the coupler is restarted, the modification done is lost therefore you should use a file "startintip" to request the modification during the coupler start-up.

- For GD/GA coupler:
  It could be change by connecting to GD/GA board:
  - [root@eMGD admin]# monitor
  - monitor_bg: no process killed
  - Start monitor using /dev/rtf1 and /dev/rtf2
  - MG:mcc
  - MCC:jitterdepth
  - Max Jitter Sz (ms) :64 <-- Enter an Hexa value
  - MaxVoiceJitterSz 100 <-- the result is 100 ms
  - 0000006A-375C1BCA: End of command execution

  If GD/GA is reset default parameters will be anew applied. 0
  Recommended “Jitter depth” value range: 100 – 100

- For IP Phones and the 4980 application, the table shows jitter buffer size (but not jitter depth), which changes dynamically.

  In fact, the buffer size varies according to the jitter present on the network. If there is too much jitter on the network then buffer jitter size will be increased (by step of one RTP packet) else it will be decreased. The table shows current buffer size each time the DSP requests a packet. This result indicates the general quality of the network.
  The unit is the RTP framing , but to have a correspondence between the buffer size and the jitter present on the network , we must use the following formula:

  Jitter = +/- (2*size of buffer)+1) * (framing RTP) /2 -> f(x) = +/- (2*x)+1) * (framing RTP) /2
  Which gives for an RTP framing of 20 ms :

  ... ...

buffer size:	**jitter:**
0	f(0) = +/- 10 ms
1	f(1) = +/- 30 ms
2	f(2) = +/- 50 ms
...	...

High levels of jitter are not acceptable for applications running in real time. This results in signal distortion, requiring the introduction of additional delays necessary for packet reassembly. IP-Touch uses a Dynamic buffer of 12 packets maximum.
Jitter average (Jitter) calculation for this call leg (call flow of the segment) is as followed:
For x = 0 to 9, f(x) = (2x + 1) * (Framing RTP/2) Framing RTP = [field ID n°19]
Jitter = (JitterDepth0* f(0)) + (JitterDepth1* f(1)) + … + (JitterDepth9* f(9)) ( ) JitterDepth0 + JitterDepth1+ … + JitterDepth9

$$\rightarrow \overline{\text{Jitter}} = \frac{(\text{JitterDepth0}*f(0)) + (\text{JitterDepth1}*f(1)) + \ldots + (\text{JitterDepth9}*f(9))}{\text{JitterDepth0} + \text{JitterDepth1} + \ldots + \text{JitterDepth9}} \qquad \overline{X} = \frac{\sum xf}{\sum f}$$

With JitterDepthx = number of measurements for size buffer x (x=0 to 9)
-> 4980 Softphone IP (PCMM2), identical to IPPhone, except in G729 where it used 10msec as unit
eVA: filled only if there had been a recording phase -> not sensitive to network burst, only loss of packets and loss of sequence.

3.3.2.10 Fields not usefull for VoIP QoS :

Field [33]: ICMP

Number of unreachable destination ICMP frames received.

Field [34]: [Fax] Number of packets lost in V21 et T4 phase

This field is not treated for the moment.

Field [35]: [Fax] number of Underrun in T4 phase

This table of 4 counters shows the number of underrun errors during a T4 phase. At the end of every T4 phase of fax communication, if the number of underrun is in one of the following intervals [1..6], [6..11], [11..16] and [16..+infini] then the corresponding counter is incremented by 1. For example, if during a T4 phase nine underrun errors were noted then the 3rd counter will be incremented by 1.

Field [36]: [Fax] number of Underrun in V21 phase

The counter is incremented by 1 each time one or many underruns are produced during a V21 phase.

3.3.2.11 VoIP parameter that could be used to calculate VoIP QoS (only for IPTouch):

Field [37]: Network jitter distribution

`[0]:0 | [1]:0 | [2]:0 | [3]:0 | [4]:0 | [5]:0 | [6]:0 | [7]:0 | [8]:0 | [9]:0`

This field describes the Jitter depth distribution, which in the case of INTIP/GD/GA will give the distribution of the delay in the jitter buffer, introduced by a jitter in the network.

In the case of IPPhone/4980 it will rather give the network distribution.

This parameter is supposed to give the network distribution for those equipments which gives the delay distribution (refer to field [32]).

For each `DSP_FRAMING_DURATION` (refer to field [41]), the number of packets received from the network since last time will be added in the table. If 3 packets have been received, the column “3” will be incremented by one.

3.3.2.12 Miscellaneous information:

Field [38]: Software version

Soft ware Version of firmware working on the coupler or IP phone

Example:

The version 2.05.0 would be coded “0x02 0x05”

The version 4.18.0 would be coded “0x04 0x12”

The version 13.02.1 would be coded “0x0d 0x02” (the “.1” is not indicated)

eVA : always coded as “0x01 0x00”

Field [39]: Terminal MCDU<IPP,eVA<

If the terminal that transmitted the ticket is an IP phone (IPP or IPT) or eVA, this field shows the directory number (MCDU) of this IP equipment.

This field is added by the Call Server when receiving the ticket at the end of the communication. The OmniVista 8770recovers data concerning IP equipments, so it will use the IPP_MCDU to find tickets for a specific IP terminal and `LOCAL_IP_ADDRESS` for the gateways.

Field [40]: Network Number

Local network number of IP equipment that transmitted the ticket.
This field is filled by CPU during reception of the message.

3.3.2.13 Parameters on RTP packet flow that could be used to calculate VoIP QoS:

Field [41]: DSP Framing Duration

The frequency in milliseconds used with the DSP for giving /receiving frames.
It could be different from the framing network, It depends on used compression algorithm and used DSP.

INTIP/GA/GD (independent of IP framing):

- G711=20 ms
- G729=20 ms
- G723=30 ms

IPPhone (independent of IP framing):

- G711=10 ms
- G729=10 ms
- G723=30 ms

IP Softphone (independent of IP framing):

- G711=10 ms
- G729=10 ms
- G723=10 ms

Field [42]: NB SID Transmitted

Refer to field [26].
This information is presented in the ticket to improve the calculation for traffic sent to the equipment which does not generate the IP statistics ticket.

Field [43]: Data transparency

A G711 communication passed as data transparency (decision by the CPU).
This information is added by the coupler.

Field [44]: Modem Transparency

A G711 communication passed as modem transparency (permission from the CPU, decision by the coupler).
This information is added by the coupler.

Field [45]: Min Delay

Minimum network delay calculated on a two way for this IP segment.

3.3.2.14 Fields about network configuration:

Field [46]: use of 802.1Q

Field that indicates if the IP equipment use or not priority 802.1Q and VLAN ID.

- **true (1)**: used
- **false (0)**: not used

Field [47]: 802.1Q Priority

Field that indicates which priority is used by IP equipment if field [46] is “true” (802.1Q is used).

Field [48]: VLAN id

Virtual LAN-id, see IEEE802.

Field that indicates which VLAN ID is used/tagged by IP equipment if field [46] is “true” (802.1Q is used).

Field [49]: Diffserv

Differentiated Services CodePoint (DSCP) MAO Value to be used for tagging the IP frames in “precedence” or “Differentiated Services CodePoint” use.

The value occupies the 6 MSB in the TOS field in IP frame. The two LSBs are CUs (Currently Unused).

Fields [50] to [54] are not used

Field [55]: Encryption

This field shows whether or not the call is encrypted (protected) by an IPTouch Security Module (e.g. MSM):

- **true (1)**: encrypted
- **false (0)**: not encrypted

This could create an additional delay or a sort of blank communication (just noise) if the encryption is not correct.

Fields [56] to [59] are not used

Field [60]: Local Jetlag

IPView and the OmniVista 8770 use the local data systems (UNIX/NT) to decode the date of an IP ticket .This date is equal to the number of seconds elapsed since 1st January 1970 and coded in 4 octets. For example, an IP ticket is sent to England at 9am. If we decode the IPticket in London, IPView shows 9am . On the contrary , if we decode the same ticket in Colombo , IPView shows 10 am. Therefore, an information (Local Jetlag) was added to the call Handling level on TLV in the IP ticket which will enable finding for any country, any date (summer or winter time) the local date of PABX which sent the IP ticket.

This information is the difference in time between local time of the PABX and GMT (+/- 12 hours).

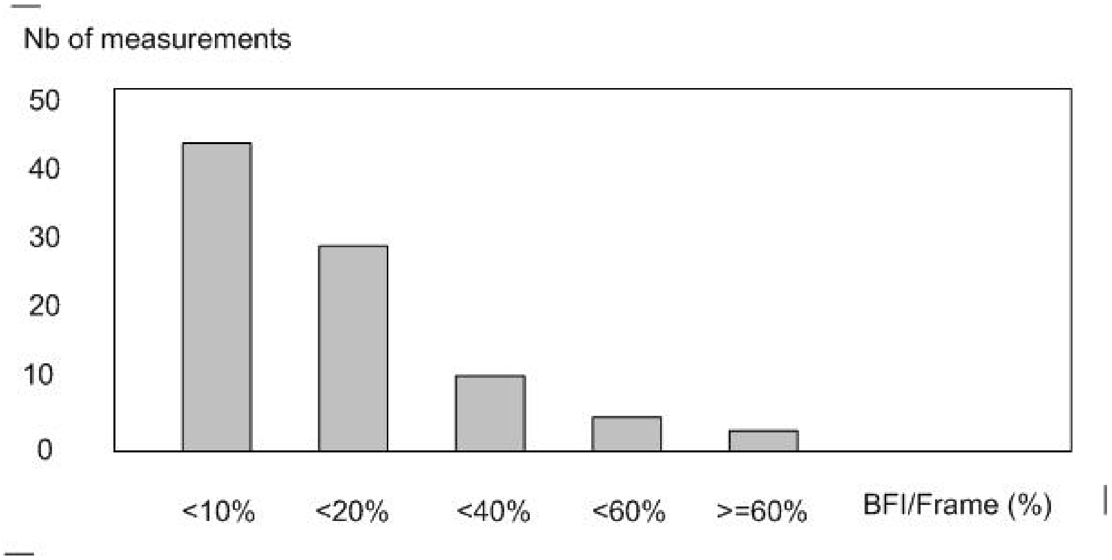
3.3.2.15 VoIP parameters that could be used to calculate VoIP QoS:

Field [61]: bfi distribution over 200ms

`[<10%]:0 | [<20%]:0 | [<40%]:0 | [<60%]:0 | [>=60%]:0 |`

This table shows BFI/density on 200ms time intervals (BFI / frame, the silence is not taken into account).

This only applies to the IP Touch set.



Values correspondences:

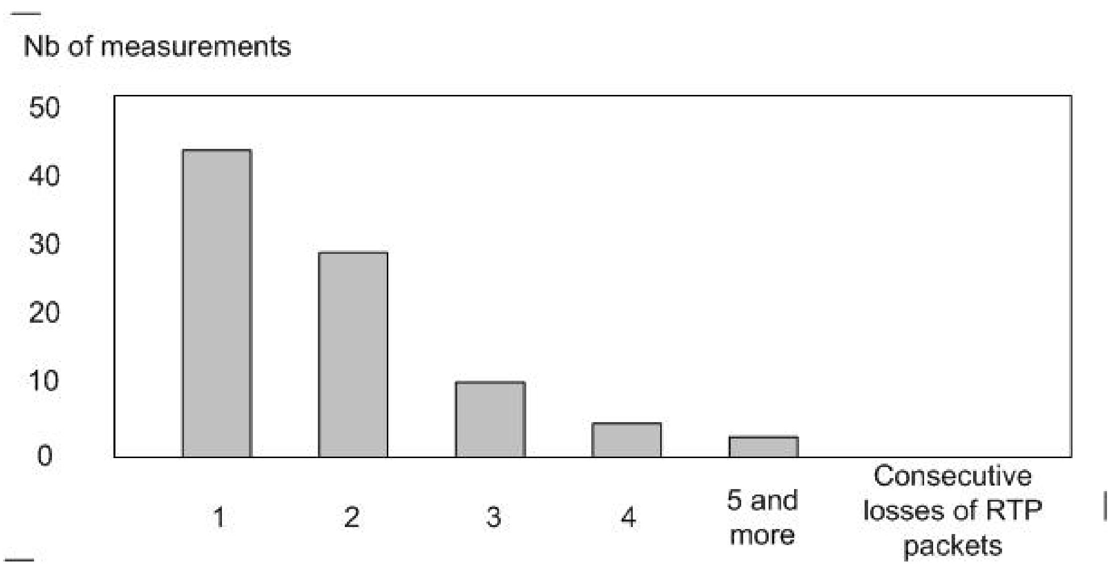
0	< 10%
1	< 20%
2	< 40%
3	< 60%
4	>= 60%

Field [62]: RTP consecutives lost

`[1]:0 | [2]:0 | [3]: 0 | [4]:0 | [5et+]:0 |`

This table of 5 values shows the occurrence of consecutive RTP packet losses.

This only applies to the Alcatel-Lucent IP Touch 4018 phone Extended Edition and Alcatel-Lucent IP Touch 4028/4038/4068 sest.



Values correspondences:

0	1
1	2
2	3
3	4
4	>=5

3.3.2.16 *Specific fields for debug:*

The debug specific fields to be implemented begin on 0x50 and a common list of different equipments will be kept updated daily

Field [80]: Transit

This parameter shows whether or not the call is in transit on the INT-IP board. Transit is configured via the parameter Transit on IP boards in the section IP > IP Parameters. This function allows RTP flows between gateways to be optimized and decompression/compression operations with a negative impact on voice quality to be avoided.

Field [81]: coder restart number

Counter incremented at every restart of the DSP.

Field [82]: Number of Ethernet driver restart

Counter incremented at each restart of Ethernet driver.

Field [83]: Problem on HDLC driver

Field indicating a serious problem on the QMC driver. Each value taken corresponds to a particular problem whose list will be given later.

Field [84]: communication type

Field indicating if the communication is of h323 Gateway type or remote/IP-Phone type.

Field [85] DSP: Estimated ERL

The new High Performance Echo Canceller in the GIP4/MADA cards may give an estimation of the echo level.

Field [86]: DSP - Estimated Delay

The new High Performance Echo Canceller in the GIP4/MADA cards may give an estimation on the echo time.

Field [87]: DSP - Background noise

The new high performance Echo Canceller in the GIP4/MADA cards can give information about the background noise. This information may be of interest to Rand D

Field [89]: Buffer depth transparency

A communication in G711 which passes as transparency needs a buffer to compensate the network jitter. The depth of buffer can be controlled by MAO. Placing a high value will avoid some failures due to loss of packets, but may cause a failure due to extra time added by the buffer.

Field [90]: Transparency of number of dummy packets run

A communication on G711 which passes as transparency needs a buffer to compensate the network jitter. If the buffer is not large enough to fill the jitter, DSP will receive a dummy version instead of a real packet `T_TRANS_PACKET_SILENCE 0x5a (90)`.

3.3.2.17 Ticket example for IP Tickets

Ticket view example as seen from “ipview” tool

Note:

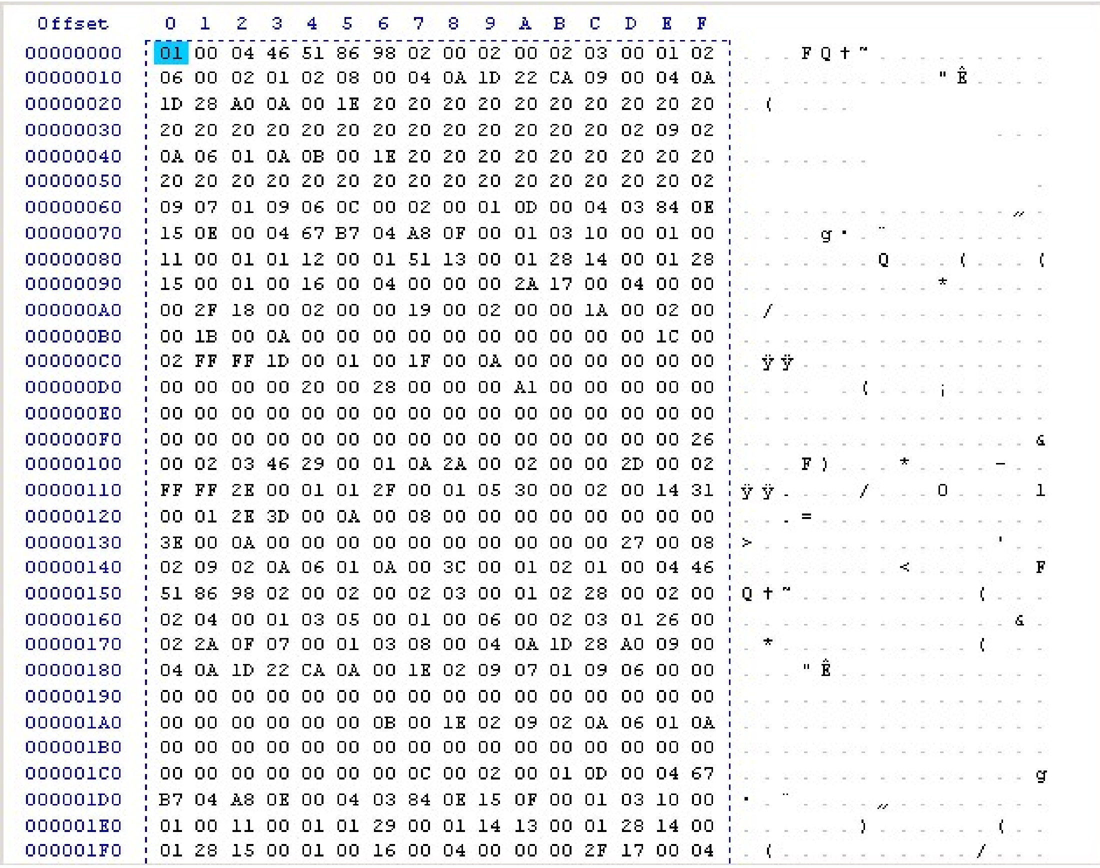
figure: IP Ticket Example shows the results for the first 60 parameters.

```
[ 1] End of communication          : Mon Mar 26 09:50:54 2007
[ 2] Node Number                   : 1
[ 3] Protocol Version              : 2
[ 4] Crystal Number                : 3
[ 5] Coupleur Number               : 0
[ 6] Equipement Type               []=     2 values
       [Type:1=IPP|2=APC|3=CplOmEnt|4=CplOmOFF] : 3
       [Type:1=IPPv2|2=NOEIPP|0=4980|1=WSftIP|0=IntIP|1=GD|2=eVA] : 1
[ 7] Nb Timeslot                   : 3
[ 8] Local IP                      : 172.25.33.13
[ 9] Distant IP                    : 172.25.35.138
[10] Local ID                      : 32000
[11] Distant ID                    : 35002
[12] Call Duration                 : 7 sec
[13] Local SSRC                    : 0x797a5d8b
[14] Distant SSRC                  : 0x0
[15] Algo Compression Type        0=G711A|1=G711U|2=G723|3=G729: 0
[16] VAD                           : false
[17] Echo Canceler                 : true
[19] Requested Framing Duration    : 30 ms
[20] Received Framing              : 0 ms
[21] Framing Change NB             : 0
[22] RTP Received Paquets NB       : 0
[23] Total RTP Paquets Sent        : 0
[24] RTP Lost Paquets NB           : 0
[25] Total Silence                 : 0 sec
[26] NB SID Received Paquets       : 0
[29] NB DTMF Detected              : 0
[30] Consecutive BFI               []=    10 values
       [1] : 0  | [2] : 0  | [3] : 0  | [4] : 0  | [5] : 0  |
       [6] : 0  | [7] : 0  | [8] : 0  | [9] : 0  | [10] : 0 |
[31] Distribution BFI              []=     5 values
       0 | 0 | 0 | 0 | 0 |
[32] Jitter Depth                  []=    10 values
       [0] : 0 | [1] : 0 | [2] : 0 | [3] : 0 | [4] : 0 | [5] : 0 |
       [6] : 0 | [7] : 0 | [8] : 0 | [9] : 0 |
[33] ICMP                          : 12
[35] FAX Underrun T4               []=     4 values
       [1-6] : 0   | [6-11] : 0  | [11-16] : 0 | [+16] : 0   |
[36] FAX Underrun V21              : 0
[37] Network Jitter Distributiont  []=    10 values
       [0] : 0 | [1] : 0 | [2] : 0 | [3] : 0 | [4] : 0 | [5] : 0 |
       [6] : 0 | [7] : 0 | [8] : 0 | [9] : 0 |
[38] Software Version              []=     2 values
       42 | 17 |
[40] Network Number                : 0
[41] DSP Framing Duration          : 20 ms
[42] NB SID Transmitted            : 0
[43] Data  transparency            : false
[44] Modem transparency            : false
[46] Qos 8021Q Used                : 0
[47] Qos 8021Q Priority            : 0
[48] Qos Vlan Id                   : 0
[49] Qos Diffserv (DSCP)           : 46
[55] Ticket Encryption             : false
[60] Local Jetlag                  : 2
```

Figure 3.7: IP Ticket Example

3.3.3 Binary ticket reading

VoIP Tickets data file (uncompressed file *.DAI) uses TLV (Type-Length-Value) format (Type=1 Byte, Length=2 Byte, Value). This file can be displayed with a Hexa editor:



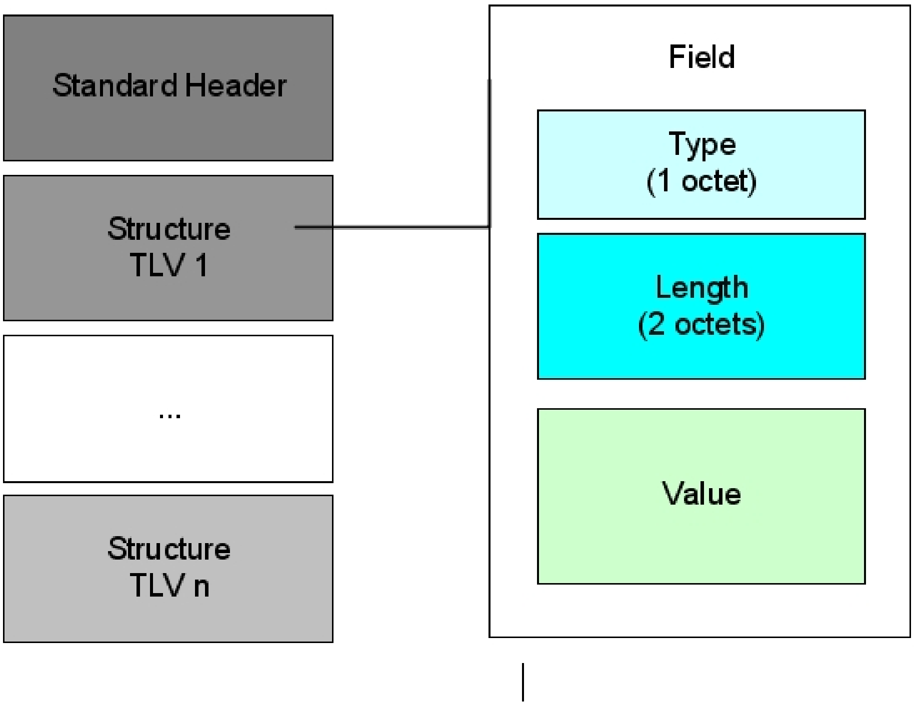
Offset	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
00000000	01	00	04	46	51	86	98	02	00	02	00	02	03	00	01	02
00000010	06	00	02	01	02	08	00	04	0A	1D	22	CA	09	00	04	0A
00000020	1D	28	A0	0A	00	1E	20	20	20	20	20	20	20	20	20	20
00000030	20	20	20	20	20	20	20	20	20	20	20	20	20	02	09	02
00000040	0A	06	01	0A	0B	00	1E	20	20	20	20	20	20	20	20	20
00000050	20	20	20	20	20	20	20	20	20	20	20	20	20	20	20	02
00000060	09	07	01	09	06	0C	00	02	00	01	0D	00	04	03	84	0E
00000070	15	0E	00	04	67	B7	04	A8	0F	00	01	03	10	00	01	00
00000080	11	00	01	01	12	00	01	51	13	00	01	28	14	00	01	28
00000090	15	00	01	00	16	00	04	00	00	00	2A	17	00	04	00	00
000000A0	00	2F	18	00	02	00	00	19	00	02	00	00	1A	00	02	00
000000B0	00	1B	00	0A	00	00	00	00	00	00	00	00	00	00	1C	00
000000C0	02	FF	FF	1D	00	01	00	1F	00	0A	00	00	00	00	00	00
000000D0	00	00	00	00	20	00	28	00	00	00	A1	00	00	00	00	00
000000E0	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
000000F0	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	26
00000100	00	02	03	46	29	00	01	0A	2A	00	02	00	00	2D	00	02
00000110	FF	FF	2E	00	01	01	2F	00	01	05	30	00	02	00	14	31
00000120	00	01	2E	3D	00	0A	00	08	00	00	00	00	00	00	00	00
00000130	3E	00	0A	00	00	00	00	00	00	00	00	00	00	27	00	08
00000140	02	09	02	0A	06	01	0A	00	3C	00	01	02	01	00	04	46
00000150	51	86	98	02	00	02	00	02	03	00	01	02	28	00	02	00
00000160	02	04	00	01	03	05	00	01	00	06	00	02	03	01	26	00
00000170	02	2A	0F	07	00	01	03	08	00	04	0A	1D	28	A0	09	00
00000180	04	0A	1D	22	CA	0A	00	1E	02	09	07	01	09	06	00	00
00000190	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
000001A0	00	00	00	00	00	00	0B	00	1E	02	09	02	0A	06	01	0A
000001B0	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
000001C0	00	00	00	00	00	00	00	0C	00	02	00	01	0D	00	04	67
000001D0	B7	04	A8	0E	00	04	03	84	0E	15	0F	00	01	03	10	00
000001E0	01	00	11	00	01	01	29	00	01	14	13	00	01	28	14	00
000001F0	01	28	15	00	01	00	16	00	04	00	00	2F	17	00	04	

Figure 3.8: Binary IP Ticket Example

To know more about third party applications and reading the binary file, contact the Alcatel-Lucent contact

3.3.3.1 *How to read binary file*

Third party applications must read ticket binary file (uncompressed “*.DAI” file) as specified below in order to prevent ticket format update.



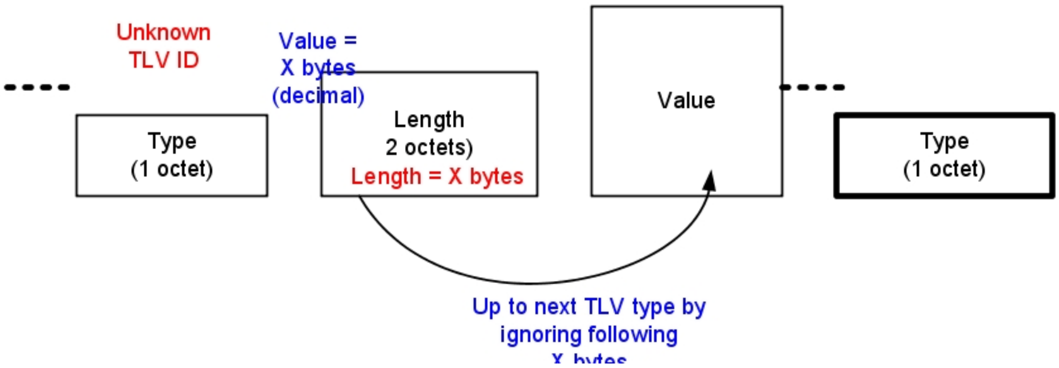
The first three TLVs are always in increasing order starting at the beginning of the binary file.

a. Ticket file reading process
Third party application must read:

- the first TLV type (first byte) then following 2 bytes that indicates value length (i.e.: X bytes) and then the value (on following X bytes).
- the second TLV type (that is following above value) then following 2 bytes that indicates value length (i.e.: X bytes) and then the value (on following X bytes).
- ... do the same for following TLV type
  The TLV type indicate the TLV ID (see table 6.1.1) The first three “TLV ID” and respectively their length will never change (first bytes must be: 01 00 04 <value> 02 00 02 <value> 03 00 01 <value>).

b. TLV type unknown
If a TLV type is not known or not take into account by Alcatel-Lucent Partner this one must be discarded when 3rd application will read the ticket file:

- Read TLV type unknown
- Read length value (X bytes) of this unknown TLV type on following 2 bytes
- Go up to next TLV type by a hop of X bytes
- Continue “Ticket file reading process” [a)]



Respect of these processes allow to 3rd application to read always correctly ticket files even if an update is done on this one by OXE party (i.e. : new field (TLV type) added, length modification of a field, ...)

Increase of "Protocol version" field will indicate a major format ticket update. At this time protocol version value is "2" (decimal). In case of Protocol version increase (higher than 2) refer to your Alcatel-Lucent contact to know the change that have been done.

3.3.3.2 How to trace a communication

The hour in the ticket is the local hour of the Communication Server which produced the ticket. It is mandatory to synchronize all the Communication Server network with the same date and time to retrace correctly a specific communication by using the "End of communication" field [1]. The Communication Server synchronization can be done via NTP (Network Time Protocol) from a reference clock (NTP server).

Every ticket contains some information to trace a communication from end to end.

The relative information includes:

- The local SSRC (identifies the IP segment)
- The distant SSRC (identifies the IP segment)
- Date of the end of communication (given by CPU)
- Duration of communication
- Local ID (identifies calling/called party
- Distant ID (identifies calling/called party)
- Local IP (identifies end-points of the IP segment)
- Distant IP (identifies end-points of the IP segment)

Note:

The Date of communication start = "Date of end of communication" – "Call duration"

On correlation context "Communication" means a direct call done between two parties: when an operation such as "on hold", or conference is involved it will be considered as a new communication. It is not possible to correlate a complete call that includes an operation like "on hold" during the same call.

3.3.3.3 Correlation algorithm

The following process is used to find the correlation between tickets of the same communication:

1. Join all tickets between “date of end of communication”.
2. Join couple of IP tickets of the same segment thanks to Local SSRC and Distant SSRC fields:
   - For a ticket with local SSRC = X and distant SSRC = Y search its symmetric ticket in which local SSRC = Y and/or distant SSRC = X on all Communication Servers network
   - List tickets by pair (2 tickets for a segment)
3. Join all segments per “date of end of communication” in ascending order in order to have all possible **segments of the same communication.**
4. **Ringing back tone ticket can be deleted** (because it is not useful for QoS analyse): Tickets generated for ringback tone can be identified by comparing local ID and Distant ID on the 2 symmetric tickets: if local ID and distant ID are the same on the 2 symmetric tickets and if ticket call duration is less than 10 seconds it can be deduced that this segment is used for ringback tone.
5. Correlate IP segments of the same communication: - .
   - Select the first IP segment (reference)
   - Check for IP segments that have the same “date of end of communication” +/- 5 seconds (*) if at least one of fields Local ID or Distant ID are the same than Local ID or Distant ID of IP segment reference in order to know if these segments are part of the same call.

   Do the same for following IP segments

   Note:
   Segments that are already correlated do not need to process this step.
6. Display IP path between end points by using Local IP, Distant IP (i.e.: IPN1 – GDN1 – GDN2 – IPN2)

(*)This time range is take into account because of delay process (signalling exchange) that could have between several nodes (OXE) for ticket generation at each end-user

3.3.3.4 Call direction

The initiator of a call can only be determined on a call where the IP ticket is generated for ringback tone. “Ringback tone” tickets generated on both sides the “Local ID” field will be equal to calling party MCDU and the “Distant ID” field will be equal to called party MCDU.