



MAF ICIMS™ – Glossary of Terms

Monitoring and Reporting for Skype for Business



Summary

Report Designer

Rates

Carrier – The name of the company which provides telecommunication services.

Cost – Cost of the services provided by telecommunication entity.

Cost 2 - Cost of the services provided by telecommunication entity.

Destination Type – Can be set up as International, International-Mobile, Local or can be personalised based on prefixes.

Call Details

Account – a number (group of numbers) which will be dialled in front of telephone number. (i.e. a lawyer's company has an account for a certain client to be able to bill all the charges made for client's purpose.

Call ID – A unique identifier for every call.

Call Type – Call type name abbreviated

Call Type Name – Type of the call like Abandoned, Busy, Conference.

Channel – A gateway can have more channels.

Conf. Organiser - The sip address of the conference organiser.

Conference ID – An identifier which allows you to follow the call chain.

Data Source – The set up and configuration method of collecting CDRs from Skype for Business.

Date - Date on which call took place.

Day - Day on which call took place.

Dialled Number - Depending on context that field can be either the external number dialling in(CLID) or the number that a user dialled.

Direction - Defines weather or not the call was incoming, outgoing or Internal.

Duration - Duration of time that call was live from the moment it was picked up and until it ended.

Extension – Extension number.

Extension Type – Can be cellular, phone, fax or sip.

Extra string 1 – Personalised parameters can be added.

Extra string 2 – Personalised parameters can be added.

Extra string 3 – Personalised parameters can be added.

Gateway - Is a network node that connects two networks using different protocols together.

Referred by – Is a field in CDRs which contains information regarding the call chain, usually it's the extension that passed the call via a transfer.

Ring time - Total time call rang before connection or disconnection.

Service type - This is the call modality (IM / App share / Voice / Video / Data)

Time - The Time the call took place.

Call Types

Is app. Sharing – if the users are sharing their screens it will show 'Y'.

Is Conference – 'Y' will appear in the column if the call is conference or 'N' if it isn't.

Is Federated - 'Y' will appear in the column if the call is federated or 'N' if it isn't.

Is File Transfer - 'Y' will appear in the column if the call contains file transfers.

Is Response Group - 'Y' will appear in the column if the call is response group call or 'N' if it isn't.

Destination

Location – Where geographically the call came from.

Phone – the location based on the dialled number.

Phone Group – A group of destinations.

Region – Continents.

Employee

Employee – Name and sip address of the employee.

Employee first Name - Users First name.

Employee ID - Users Sip address.

Employee Last Name - Users Last Name.

Employee name - Users First and Last Name.

Extension location – Location of the extension from which the call was made.

Hierarchy

Ancestor Unit – In multilevel hierarchy, i.e. Company->Town->Department -> Employee (ancestor unit of the employee is town and organisation unit is Department)

Organisation unit - Assigned department within organisation

IP Fields

Connection Type – The connection type that user is using, i.e. Ethernet, WI-FI, Wired.

Dest. Audio codec – Is a codec (a device or computer program capable of encoding or decoding a digital data stream) that encodes or decodes audio.

Dest. Resolution - Video calls resolution

Dest. Video codec – Is an electronic circuit or software that compresses or decompresses digital video. It converts uncompressed video to a compressed format or vice versa.

Destination IP – Internet Protocol address is a numerical label of the recipient, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Destination IP v6 - Is the most recent version of the Internet Protocol (IP), the communications protocol that provides an identification and location system for computers on networks and routes traffic across the Internet.

From reflexive local IP – Reflexive local IP contain condition statements (entries) that define criteria for permitting IP packets.

Jitter – Measures the variability of packet delay and results in a distorted or choppy audio experience. Jitter can increase latency on networks.

Packets - A network packet is a formatted unit of data carried by a packet-switched network.

Latency – Is a time interval between the stimulation and response, or, from a more general point of view, a time delay between the cause and the effect of some physical change in the system being observed.

MOS – Mean opinion score – is the gold standard measurement to gauge the perceived audio quality

Can be between 1 and 5: - 1 (Bad)
 - 2 (Poor)

- 3 (Fair)
- 4 (Good)
- 5 (Excellent)

Octet - Is a unit of digital information in computing and telecommunications that consists of eight bits. The term is often used when the term byte might be ambiguous, as the byte has historically been used for storage units of a variety of sizes.

Octets received – The amount of received octets.

Octets sent – The number of octets sent.

Orig. audio codec – Is a codec (a device or computer program capable of encoding or decoding a digital data stream) that encodes or decodes audio.

Orig. video codec – Is an electronic circuit or software that compresses or decompresses digital video. It converts uncompressed video to a compressed format or vice versa.

Originator IP – Internet Protocol address is a numerical label of the remitter, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Originator IPv6 – Is the most recent version of the Internet Protocol (IP), the communications protocol that provides an identification and location system for computers on networks and routes traffic across the Internet.

Packets lost – packet Loss (%) represents the % of packets that did not make it to their destination. Packet loss will cause the audio to be distorted or missing (on the receiver end).

Packets Received – The amount of received packets.

Packets sent – The amount of sent packets.

Pool – Is a set of resources that are kept ready to use, rather than acquired on use and released afterwards.

Quality – Is an international standard, developed by Virtual Socket Interface Alliance for measuring IP or SIP (Silicon intellectual property) quality and examining the practices used to design, integrate and support the SIP.

Server – Is a computer program or a device that provides functionality for other programs or devices, called "clients". These can be either physical or virtual machines.

SIP response code – Is a signalling protocol used for controlling communication sessions such as Voice over IP telephone calls. SIP is based around request/response transactions, in a similar manner to the Hypertext Transfer Protocol (HTTP). Each transaction consists of a SIP request (which will be one of several request methods), and at least one response.

Subnet – Is a logical subdivision of an IP network. The practice of dividing a network into two or more networks is called subnetting.

Subnet location – Location of the subnetwork.

To reflexive local IP – Reflexive local IP contain condition statements (entries) that define criteria for permitting IP packets.

VPN – A virtual private network extends a private network across a public network, and enables users to send and receive data across shared or public networks as if their computing devices were directly connected to the private network

Skype

App. Sh. Avg. jitter – Measures the variability of packet delay and results in a distorted or choppy audio experience. Jitter can increase latency on networks.

Avg. Net MOS – Network MOS is a prediction of the wideband Listening Quality Mean Opinion Score (MOS-LQ) of audio that is played to the user. This value takes into consideration only network factors such as codec used, packet loss, packet reorder, packet errors and jitter.

Call admission control – Prevents oversubscription of VoIP networks. It is used in the call set-up phase and applies to real-time media traffic as opposed to data traffic.

Callee – The agent / employee / user receiving a call.

Callee app. Sh. Relative one-way avg. – Optimal value for the relative one-way delay between the two media endpoints involved in the application sharing. This is a single-hop latency measure

Callee app. Sharing bandwidth (Kbps) – Is a type of resource allocation designed to model the real-world allocation of bandwidth to many users in a network.

Callee audio bandwidth (Kbps) – This refers to the frequency range a device can carry without degrading any of the information. It's also used in digital communication to show how much information something can transfer over a given time.

Callee audio packets lost rate – Packet Loss (%) represents the % of packets that did not make it to their destination. Packet loss will cause the audio to be distorted or missing (on the receiver end).

Callee audio round trip – Is the most common measure of latency and is measured in ms.

Callee avg. jitter – Measures the variability of packet delay and results in a distorted or choppy audio experience on the receiving end.

Callee avg. listening MOS – Is a prediction of the wideband Listening Quality (MOS-LQ) of the audio stream that is played to the user.

Callee avg. MOS – Average means opinion score.

Callee avg. net MOS degradation - Network MOS Degradation for the whole call. This metric shows the amount the Network MOS was reduced because of jitter and packet loss

Callee avg. sending MOS – Is a prediction of the wideband Listening Quality Mean Opinion Score (MOS-LQ) of the audio stream that is being sent from the user. This value takes into consideration the speech and noise levels of the user along with any distortions, and from this data predicts how a large group of users would rate the audio quality they hear.

Callee client type – Client type, i.e. Skype for Windows, Skype for iPhone, Skype for Android.

Callee client version – Client version.

Callee conv. MOS – Is a prediction of the narrowband Conversational Quality (MOS-CQ) of the audio stream that is played to the user. This value takes into consideration the listening quality of the audio played and sent across the network, the speech and noise levels for both audio streams, and echoes. It represents how a large group of people would rate the quality of the connection for holding a conversation.

Callee dynamic capability % - Percentage of the call where the client experienced high CPU load when processing video

Callee echo mic in – Echo that was present in the microphone. Typically, you will see low values for headsets or handsets, and higher values for speaker phones or stand-alone speakers.

Callee echo send – Echo transmitted to other users on the call.

Callee end point – Is a device or node that is connected to the LAN or WAN and accepts communications back and forth across the network.

Callee inbound video frame rate avg. – The average video frame rate received during the call.

Callee low frame rate call % - Percentage of low frame rate call.

Callee low network BW – Is the minimum rate of data transfer across a given path. Bandwidth may be characterised as network bandwidth, data bandwidth, or digital bandwidth.

Callee max jitter – Measures the maximum variability of packet delay and results in a distorted or choppy audio experience.

Callee max net MOS degradation – This metric shows the maximum amount the Network MOS that was reduced because of jitter and packet loss.

Callee MIC. not functioning – Calls in which the capture device was not functioning at an acceptable level. A high value suggests that quality issues with the call were primarily due to the capture device not working as expected.

Callee min net MOS – This metric shows the minimum amount the Network MOS was reduced because of jitter and packet loss.

Callee near end to echo – Used in telephony to improve voice quality by preventing echo from being created or removing it after it is already present.

Callee network connection – Network connection.

Callee outbound video frame rate avg. – The average video frame rate sent during the call.

Callee PAI – P-Asserted-Identity.

Callee ratio concealed samples avg. – Concealing audio samples is a technique used to deal with network packets.

Callee RDP tile processing latency avg. – Acceptable value of the average RDP tile processing latency in the AS Conferencing Server over the duration of the viewing session.

Callee recv. frame rate avg. – Average video frame rate used by the receiver

Callee render device – Device (for example, a headset or speakers) used for receiving audio.

Callee spk. not functioning – Calls in which the render device was not functioning at an acceptable level. A high value suggests that quality issues with the call were primarily due to the render device not working as expected.

Callee spoiled tile % total – Total percentage of spoiled RDP tiles

Callee subnet – The subnet the callee resides on.

Callee URI – A Uniform Resource Identifier is a string of characters used to identify a resource.

Callee video avg. jitter – Average jitter in video calls.

Callee video bandwidth (Kbps) – Video calls bandwidth.

Callee video local frame loss % avg. – The percentage of total video frames that are lost.

Callee video packets loss rate – The packet loss rate for video calls.

Callee video post FECPLR – The packet loss rate after forward error correction has been applied.

Callee video round trip – Round trip time for video calls.

Callee voice switch – Calls which had to be placed into half duplex mode. In half duplex mode, communication can travel in only one direction at a given time.

Callee VPN – A virtual private network extends a private network across a public network and enables users to send and receive data across shared or public networks as if their computing devices were directly connected to the private network.

Caller – The agent / Employee / user making a call.

Caller app. Sh. Relative one-way avg. – Optimal value for the relative one-way delay between the two media endpoints involved in the application sharing. This is a single-hop latency measure

Caller app. Sharing bandwidth (Kbps) – Is a type of resource allocation designed to model the real-world allocation of bandwidth to many users in a network

Caller audio bandwidth (Kbps) – This refers to the frequency range a device can carry without degrading any of the information. It's also used in digital communication to show how much information something can transfer over a given time.

Caller audio packets lost rate – Packet Loss represents the number of packets that did not make it to their intended destination. Packet loss will cause the audio to be distorted or missing.

Caller audio round trip – Is the most common measure of latency and is measured in MS.

Caller avg. jitter – Measures the variability of packet delay and results in a distorted or choppy audio experience.

Caller avg. listening MOS – Is a prediction of the wideband Listening Quality (MOS-LQ) of the audio stream that is played to the user.

Caller avg. MOS – Average Means Opinion Score.

Caller avg. net MOS degradation – Average network MOS degradation is an integer represents the amount of the MOS value lost to network affects.

Caller avg. sending MOS – Is a prediction of the wideband Listening Quality Mean Opinion Score (MOS-LQ) of the audio stream that is being sent from the user. This value takes into consideration the speech and noise levels of the user along with any distortions, and from this data predicts how a large group of users would rate the audio quality they hear.

Caller capture device – The Microphone or recording device use to capture audio.

Caller client type – Client type, i.e. Skype for Windows, Skype for iPhone, Skype for Android.

Caller client version – Client version.

Caller conv. MOS – Is a prediction of the narrowband Conversational Quality (MOS-CQ) of the audio stream that is played to the user. This value takes into consideration the listening quality of the audio played and sent across the network, the speech and

noise levels for both audio streams, and echoes. It represents how a large group of people would rate the quality of the connection for holding a conversation.

Caller dynamic capability % - Percentage of the call where the client experienced high CPU load when processing video.

Caller echo mic in – Echo that was present in the microphone.

Caller echo send – Echo transmitted to other users on the call.

Caller end point – Is a device or node that is connected to the LAN or WAN and accepts communications back and forth across the network.

Caller inbound video frame rate avg. – The average video frame rate sent during the call.

Caller low frame rate call % - Percentage of low frame rate within a call.

Caller low network BW – Is the minimum rate of data transfer across a given path. Bandwidth may be characterised as network bandwidth, data bandwidth, or digital bandwidth.

Caller max jitter – Measures the maximum variability of packet delay and results in a distorted or choppy audio experience.

Caller max net MOS degradation – This metric shows the maximum amount the Network MOS that was reduced because of jitter and packet loss.

Caller MIC. not functioning – Calls in which the capture device was not functioning at an acceptable level. A high value suggests that quality issues with the call were primarily due to the capture device not working as expected.

Caller min net MOS – This metric shows the minimum amount the Network MOS was reduced because of jitter and packet loss.

Caller near end to echo – Used in telephony to improve voice quality by preventing echo from being created or removing it after it is already present.

Caller network connection – Shows the network the caller connected to Wired / WIFI / ethernet Etc.

Caller outbound video frame rate avg. – The average video frame rate sent during the call

Caller PAI – P-Asserted-Identity.

Caller ratio concealed samples avg. – Concealing audio samples is a technique used to deal with dropped network packets.

Caller RDP tile processing latency avg. – Acceptable value of the average RDP tile processing latency in the AS Conferencing Server over the duration of the viewing session.

Caller recv. frame rate avg. – Average video frame rate used by the receiver.

Caller render device – Device (for example, a headset or speakers) used for receiving audio.

Caller spk. not functioning – Calls in which the render device was not functioning at an acceptable level. A high value suggests that quality issues with the call were primarily due to the render device not working as expected.

Caller spoiled tile % total – Total percentage of spoiled RDP tiles.

Caller subnet – The subnet the caller resides on.

Caller URI – A Uniform Resource Identifier is a string of characters used to identify a resource.

Caller video avg. jitter – Average jitter in video calls.

Caller video bandwidth (Kbps) – Video calls bandwidth.

Caller video local frame loss % avg. – The percentage of total video frames that are lost.

Caller video packets loss rate – Packet Loss represents the number of packets that did not make it to their intended destination. Packet loss will cause the audio to be distorted or missing.

Caller video post FECPLR – The packet loss rate after forward error correction has been applied.

Caller video round trip – This measure the average round-trip time for RTP packets between endpoints. If the latency is high, users are likely to hear a delay in the audio.

Caller voice switch – Calls which had to be placed into half duplex mode. In half duplex mode, communication can travel in only one direction at a given time.

Caller VPN – A virtual private network extends a private network across a public network and enables users to send and receive data across shared or public networks as if their computing devices were directly connected to the private network.

Client alias – An alias is an alternate name that can be used to make a connection. The alias encapsulates the required elements of a connection string and exposes them with a name chosen by the user.

Client version – Version of the client.

Diagnostic ID – Is a unique identifier (in the form of an ms-diagnostics header) that gets attached to a SIP message, while the Diagnostic header provides an accompanying description for the Diagnostic ID.

Disconnected by phone – the connection was interrupted due to phone issues.

Disconnected by user – The connection was interrupted due to user issues.

Error category – Type of the error occurred.

Error description – Description of the error with details.

Extension client type – Client type that the extension is using.

HD quality – High definition quality.

NMOS degradation (jitter) – Network MOS Degradation for the complete call. This metric shows the amount the Network MOS was reduced because of jitter.

NMOS degradation (packet loss) – Network MOS Degradation for the complete call. This metric shows the amount the Network MOS was reduced because of packet loss.

Pool – Is a set of resources that are kept ready to use, rather than acquired on use and released afterwards.

Rating – is a feature that provides enterprises a way to get feedback from their end-users. It offers a "star" rating system and predefined tokens for audio and video calls. In addition, administrators can enable a custom field to provide feedback.

Rating categories – "star" rating, one star means bad and five stars means excellent.

Ratio compressed samples avg. – Average ratio of compressed audio samples to the total number of samples. Compressed audio is audio that has been compressed to help maintain call quality when a dropped network packet has been detected. High values indicate significant levels of sample compression, and result in audio sounding accelerated or distorted..

Ratio stretched samples avg. – Average ratio of stretched audio samples to the total to the total number of samples. Stretched audio is audio that has been expanded to help maintain call quality when a dropped network packet has been detected. High values indicate significant levels of sample stretching caused by jitter, and result in audio sounding robotic or distorted.

SD quality – Standard quality.

Server – Is a computer program or a device that provides functionality for other programs or devices, called "clients"

Video allocated bandwidth – The amount of bandwidth that is allocated for video calls.

Video resolution – Resolution of video calls.

Response Group

Queue name – Name of the queue

Response group description – Description of the Response group

SIP address – Email address used to configure the Skype for Business account.

Telephone – Telephone.

Summary

Callee NMOS degradation* - Network MOS Degradation for the whole call. This metric shows the amount the Network MOS was reduced because of jitter and packet loss

Caller NMOS degradation * - Network MOS Degradation for the whole call. This metric shows the amount the Network MOS was reduced because of jitter and packet loss

Calls* - Outbound (out to PSTN)/ inbound (incoming from PSTN) / internal calls (internal call between Skype for business users)

Cost * - Rate associated with making or receiving calls.

Cost 2 * - Rate associated with making or receiving calls.

Duration * - Total time call was live. Picked up (connected) -> hung up (disconnected)

Extensions * - Extension number.

Jitter * - Measures the variability of packet delay and results in a distorted or choppy audio experience. Jitter can increase latency on networks.

MOS * - Mean opinion score – is the gold standard measurement to gauge the perceived audio quality

Can be between 1 and 5:

- 1 (Bad)
- 2 (Poor)
- 3 (Fair)
- 4 (Good)
- 5 (Excellent)

Ring time * - Total time call range for.

Time * - Time at which the call took place.

Report Builder

General

Date – Can be set for a specific day or range.

Time – Can be set for a specific time or range.

Duration – The elapsed time between answer and disconnect

Ring Time - Total time call rang before connection or disconnection.

Cost - Cost of the services provided by telecommunication entity.

Direction – Defines weather or not the call was incoming, outgoing or Internal.

Incoming – A call that is coming into the organisation

Outbound – An outgoing call from a user

Internal – A call that is between users within the same organisation

Service type - This is the call modality (IM / App share / Voice / Video / Data)

Voice – Audio calls.

Video – Video calls.

App. Sharing – Sharing screen during a Skype call or conference.

IM – Instant messages between Skype4b users.

Data – files transferred between Skype users

Call Types

Abandoned - The caller hung up the call without being answered. Duration of the call is 0 and ring time reflects the amount of time the call was being presented.

Start – For an internal call, Skype will generate 2 call detail records (CDRs), start leg has caller extension in Extension column and callee extension will be in CLID column.

Transfer – Agent picks up calls and transfers it out to another agent or dept.

Conference – A service feature that allows a call to be established among three or more stations in such a manner that each of the stations is able to communicate with all the other stations.

Pickup – CISCO

Tandem – i.e. A call comes outside working hours. The system can be set up to send the call to an external user or number. In the system it will appear as one incoming call and one outgoing call.

Presented – A call that has rang to an individual agent.

File transfer – Files, documents or any data transferred through Skype

Transfer out – Transferred call out to PSTN

Busy – The system closed the call depending on the configuration, i.e. call timeout, lack of voicemail or overflow system.

End – For an internal call, Skype will generate 2 call detail records (CDRs), end leg has callee extension in Extension column and caller extension in CLID column.

Forward – A call which is forwarded to another employee, department etc, action is done via an automatic system such as response groups.

Response Group – Is a feature that lets managers or server administrators route and queue incoming calls to groups of people, called agents, such as for a help desk or a customer service desk.

Park - Is a feature of some telephone systems that allows a user to put a call on hold at one telephone set and continue the conversation from any other telephone set.

Voice mail – Is a method of storing voice messages electronically for later retrieval by intended recipients.

Personal – Calls identified for a personal purpose.

Error – A call which is identified as inaccurate or incorrect

Scheduled – A call set for a specific time.

Federated – Enables a Skype for Business user to connect with users in other organisations that use Skype for Business as well as those that host their own Skype for Business Server on-premises.

CUCM Call Types

Intercom - A dedicated voice service within a specified user environment.

Barge - Enables you to drop in on live calls to speak with both the caller and the agent.

IVR - Interactive voice response is a technology that allows a computer to interact with users through the use of voice and DTMF tones input via a keypad.

Malicious – CISCO

Mobility - Calls though a mobile phone.

HandIn - CISCO

HandOut – CISCO

Cell pick up – CISCO

Call Type Abbreviations

A = Abandoned Call

B = Busy Call

X = Transferred Call

F = Forwarded Call

T = Tandem Call

S = Start Leg

E = End Leg

Err = Error Call

C = Conference Call

H = Hold Call

Pck = Pickup Call

Icom = Intervom Call

M = Mobility Call

MHin = Mobility HandIn

MHout = Mobility HandOut

CPck = Cell Pickup

IVR = IVR Call

Prk = Call Park

Mal = Malicious Call

Brg = Barge Call

Organisation Structure

Organisation Structure – The way in which employees / departments / teams are set up in AD.

Extension – Extension number (cellular / fax / phone /SIP)

Location – Where geographically the call came from.

Referred by – Is a field in CDRs which contains information regarding the call chain, usually it's the extension that passed the call via a transfer.

Employee – Name and sip address of the employee.

Department – A subset or team of users within the organisation.

Response Groups

Response Group – Is a feature that lets managers or server administrators route and queue incoming calls to groups of people, called agents, such as for a help desk or a customer service desk.

Queue name – The name assigned to a group of employees (Response Group)

All legs – Shows calls that have been transferred or bounced between several agents or response groups.

Destination

Dialled number / CLID – The number that the user call out to.

Destinations – the location/destination of the call, based on phone directory

Directory groups – directory groups can be assigned to every destination from phone directory

Destination Types – based on the destination number, destination types can be: international, national, international-mobile

Gateway

Gateway – Is a network node that connects two networks using different protocols together.

Channel – is a separate path through which signals can flow

Carriers – Company that offers communication services over land-wire, cable, mobile (cellular), point-to-point microwave, and/or satellite systems.

IP Fields

Originator IP – Internet Protocol address is a numerical label of the remitter, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Destination IP – Internet Protocol address is a numerical label of the recipient, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Subnet – A subnet (short for "subnetwork") is an identifiably separate part of an organization's network. Typically, a subnet may represent all the machines at one geographic location, in one building, or on the same local area network (LAN).

Subnet locations – Location of Subnet.

MOS – Mean Opinion Score

Quality – UCA uses the MS methodology of rate calls either GOOD or POOR quality.

Connection Type – How the call was connected between the participants. E.g.

Wired

Wi-Fi

Mobile broadband

Tunnel

VPN - Virtual Private Network

Sort and Summary – Sort and group reports by applying specific filters.

Ancestor Unit – In multilevel hierarchy, i.e. Company->Town->Department -> Employee (ancestor unit of the employee is town and organisation unit is Department)

Call type – Call types can be (see list of abbreviations above)

Carrier – The name of the company which provides telecommunication services.

Channel – is a separate path through which signals can flow

Conference ID – Each conference is given an individual ID which allows you to follow the call chain.

Conference organiser – The sip address of the conference organiser (Agent that arranged / scheduled conference).

Cost – Cost of the services provided by telecommunication entity.

Data Source – Set up and configuration method of collecting Call Detail Records (CDRs) from Skype for Business

Date – The Date on which the activity (IM / Voice / Video / Data) took place.

Destination IP – Internet Protocol address is a numerical label of the recipient, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Destination type – Can be set up as International, International-Mobile, Local or can be personalised based on prefixes.

Dialled Number – The number the user dialled

Direction – Defines whether or not the call was incoming, outgoing or Internal.

Duration – The total time between the call being picked up and disconnected.

Employee – Name and sip address of the user / agent.

Extension – Extension number.

Extension location – Location of extension geographically.

Extension Type – Can be cellular, phone, fax or sip.

Gateway – Is a network node that connects two networks using different protocols together

Month – The calendar month in which the activity took place (IM / Voice / Video / Data / App share)

Organisation unit – Assigned department within the company.

Originator IP – Internet Protocol address is a numerical label of the remitter, assigned to each device connected to a computer network that uses the Internet Protocol for communication.

Phone – carrier name or location of dialled number, based on phone directory.

Phone group – A group of destinations.

Queue name – The name assigned to a group of employees (Response Group)

Referred by – Is a field in CDRs which contains information regarding the call chain, usually it's the extension that passed the call via a transfer.

Response group – Is a feature that lets managers or server administrators route and queue incoming calls to groups of people, called agents, such as for a help desk or a customer service desk.

Ring time – Total time the call rang for, before being connected or disconnected.

Service – service type (audio, video, app. sharing, data, IM)

Subnet – A subnet (short for "subnetwork") is an identifiably separate part of an organization's network. Typically, a subnet may represent all the machines at one geographic location, in one building, or on the same local area network (LAN).

Subnet location – Location of Subnet.

Time – The time at which the activity took place (IM / Voice / Video / Data / App Share)

Week – The week at which the activity took place (IM / Voice / Video / Data / App Share)

Generate – Click to produce reports, either in the same web page or a new one.

Schedule Report – Define a frequency on which you wish the report to be run. For example, Day, Week, Month, Year. Set the time and who you wish to deliver to.

Save – The ability to save you reports to templates.

Clear – Reset the report builder to the default settings

Report Options

Format – Select the predefined report formats from the list of bespoke reports you have designed in the report designer

Currency – Select preferred currency

About MAF InfoCom™

Formed in 2000, MAF InfoCom™ is a leading innovative technology provider with almost 20 years' experience in delivering solutions for Monitoring, Analytics, Reporting and Recording of telephony and Unified Communications, Call Management, Billing & Call Accounting.

We serve tens of thousands customers around the globe, in a large variety of branches. We have installations in over 50 countries ranging from SME's to multi-national global enterprises. In Europe MAF InfoCom™ is the largest provider of UC reporting solutions.

With the market trend towards Unified Communications we expand our sales across the globe rapidly. Our solutions work with every major (IP)PBX and UC manufacturer platform.

Our solutions are offered from the Cloud, On-Premises and Partner Hosted to enable our customers and partners to choose the best model for their needs.

Monitoring, Analytics, Reporting and Recording for Unified Communications.



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