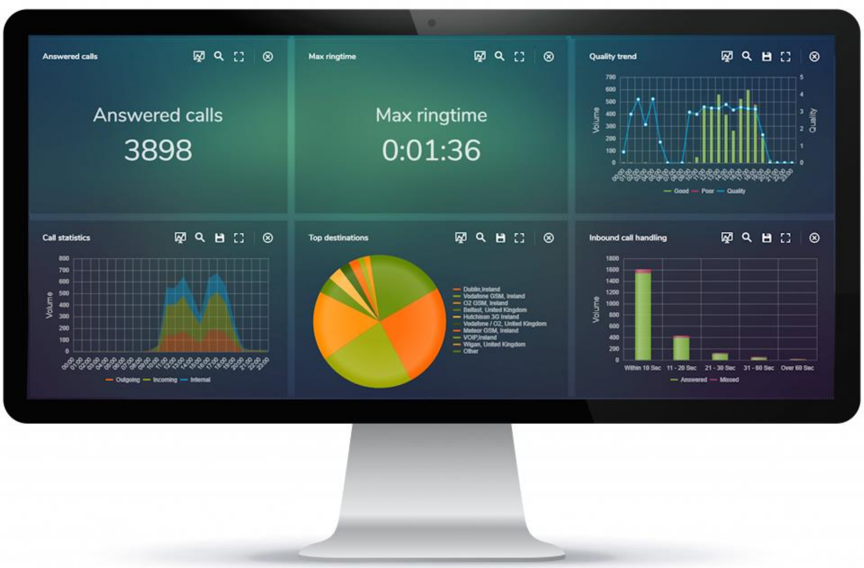
MAF ICIMS™

Glossary

Monitoring, Analytics and Reporting for UC&C



MAF InfoCom™

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 personalized based on prefixes.

Call Details

Account	...
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. Organizer	The sip address of the conference organizer.
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.
Dialed Number	Depending on context that field can be either the external number dialing in(CLID) or the number that a user dialed.
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	Personalized parameters can be added.
Extra string 2	Personalized parameters can be added.
Extra string 3	Personalized 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 dialed 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 organization unit is Department)
Organization unit	Assigned department within organization

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	…
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 signaling 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 for Business

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 characterized 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 dropped 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 characterized 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. When 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 …

Rating categories	…
Ratio compressed samples avg.	Quantify the reduction in data-representation size produced by a data compression algorithm.
Ratio stretched samples avg.	…
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 organization
Outbound	An outgoing call from a user
Internal	A call that is between users within the same organization
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 organizations 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

Organization Structure

Organization 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 organization.

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

Dialed number / CLID	The number that the user call out to.
Destinations	The location/destination on the call, based on phone directory
Directory groups	**…**
Destination Types	Can be international, national, international mobile

Gateway

Gateway	Is a network node that connects two networks using different protocols together.
Channel	**I**s 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 organization 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 induvial ID which allows you to follow the call chain.
Conference organizer	The sip address of the conference organizer (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 personalized based on prefixes.
Dialed Number	The number the user dialed.
Direction	Defines weather 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)
Organization 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	The destination of the call, 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

Who we are

Formed in 2000, MAF InfoCom™ is a leading innovative technology provider with over two decades experience delivering solutions for Unified Communications and Collaboration including Monitoring, Analytics, Reporting, Recording, Headset & Device Management and DID Management.

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 and Collaboration we expand our sales across the globe rapidly. Our solutions work with every major UC&C technology.

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.

MAF ICIMS™

UC&C Monitoring Analytics & Reporting

MAF ICIMS CC™

Live Wallboards, Real Time Agent Status

MAF NMS™

Number Management System, DID Range Management

MAF UCR™

Microsoft Teams Voice Recorder

MAF DMS™

Inventory Management for Headset and Devices

MAF QMS™

Microsoft Teams Call Queue Management System

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