CODE.

CLOBBA
Glossary

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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 –
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 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 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 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 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. 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 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 on the call, based on phone directory
Directory groups
Destination Types – **c**an 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 induvial 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 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)
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 – 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

Who we are

Code was formed in 2013 by three experts in call reporting and analytics. They identified the need at the time for a new solution which would be designed and developed from day one with consideration for the more complex reporting requirements of Unified Communications platforms.

Since our formation we have increased our product portfolio in line with our core vision to create a single application which gives visibility and management of multiple elements of a UC environment.

The **CLOBBA** suite of software includes:

Clobba RT - Real-time wallboards and trend analytics for Microsoft Teams enabling users to manage queues, improve response times, and maximize productivity

Clobba DM - showing an inventory and level of device management for headsets from EPOS, Jabra and Poly

Clobba VR - easy to use and cost-effective call recording for Microsoft Teams, our simple, compliant and affordable voice recording solution

Clobba RM - DID range, number and extension management and provisioning solution, your single pane of glass for all number-management tasks

Clobba QM - allows full configuration of Microsoft Teams Call Queues and Auto Attendants without the requirement of a Teams administrator role

The team at Code have experience of working on some of the largest installations of call reporting in the world and pride ourselves in delivering the highest levels of customer service.

We operate through a global network of 50+ partners with over 3,000 customer installations across five continents and operate from offices in UK, US, New Zealand and Romania.

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