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(54) Benævnelse: **Løsning til etablering af en kommunikationssession**

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# DESCRIPTION

## TECHNICAL FIELD

**[0001]** The invention concerns in general the technical field of telecommunication. More particularly, the invention concerns a solution for establishing a communication session between parties.

## BACKGROUND

**[0002]** The current trend in customer service is to enable interaction between a customer and a service provider remotely by using modern technologies in a field of telecommunications. Traditionally, the interaction in the remote customer service has been implemented with voice i.e. the customer e.g. calls to service provider's customer service. A more sophisticated solution is that the interaction is performed with a video call by applying a specific hardware and software application executed by the hardware for the service session.

**[0003]** A recent technology taken into use in remote communication in which both voice and video are involved in is so called web real-time communication (WebRTC) technology. The WebRTC technology allows audio or video communication to work inside web pages by allowing direct peer-to-peer communication, eliminating the need to install plugins or download native apps. Hence, the audio or video communication may be established with web browser which simplifies a user experience since an application familiar to the user may be used for voice call or video sessions.

**[0004]** An example of a utilization of the WebRTC technology is given in a document WO 2016/156256 A1. In the document it is described a solution for establishing a data communications channel between terminal devices over WebRTC communication technology Another example can be found in US2015/0029296 A1.

**[0005]** Applying the WebRTC technology for customer service solutions has a deficiency in that the WebRTC does not have tools for implementing a charging of the connection, or on the customer service. Generally speaking, it depends on the type of customer service the payer of the connection and/or the customer service itself may be the end customer or the service provider of the customer service. Hence, there is need to develop solutions which alleviate the deficiency and enables at least a generation of data on the basis of which the charging of the customer service connection may be performed.

## SUMMARY

**[0006]** The following summary merely presents some concepts of the invention in a simplified form as a prelude to a more detailed description of exemplifying embodiments of the invention.

**[0007]** An objective of the invention is to present a method and a communication system for establishing a communication session. Another objective of the invention is that at least portion of the communication session is established with web real-time communication.

**[0008]** The objectives of the invention are reached by a method and a communication system as defined by the respective independent claims.

**[0009]** According to a first aspect, a method for establishing a communication session is provided as defined in independent method claim 1.

**[0010]** The request for establishing a communication session from a first WebRTC device may comprise data for authenticating the first WebRTC device by the server.

**[0011]** Further, the method may comprise a step of generating, by the public switched telephone network, data record for storing charging data associated to the communication session established.

**[0012]** The identifier representing the first WebRTC device from which the first connection for the voice call is established may be obtained by the second WebRTC device by inquiring it from a database storing the identifier associated with the network address of the first WebRTC device, the inquiry to the database comprising a network address of the first WebRTC device obtained from the first connection request for the voice call received by the second WebRTC device. The identifier representing the first WebRTC device may be the second connection request specific and generated in response to the receipt of the acknowledgement from the server by one of the following: the first WebRTC device, the connection management server.

**[0013]** The identifier representing the first WebRTC device obtained from the database may comprise a link address to the connection management server for associating, by the connection management server, the second connection request from the first WebRTC device with the inquiry to the connection management server from the second WebRTC device.

**[0014]** Moreover, the second connection may be established with web real-time communication technology.

**[0015]** Still further, data may be carried over the second connection, the data being at least one of the following: real-time video data, video data file, document data, chat application data.

**[0016]** According to a second aspect, a communication system for establishing a communication session is provided as defined by independent system claim 9.

**[0017]** The server may be configured to acknowledge the first WebRTC device the request for establishing a communication session from the first WebRTC device to the second WebRTC device in response to an authentication of the first WebRTC device by the server based on authentication data received in the request.

**[0018]** The public switched telephone network may be configured to generate data record for storing charging data associated to the communication session established.

**[0019]** The second WebRTC device may be configured to obtain the identifier representing the first WebRTC device from which the first connection for the voice call is established by inquiring it from a database storing the identifier associated with the network address of the first WebRTC device, the inquiry to the database comprising a network address of the first WebRTC device obtained from the first connection request for the voice call received by the second WebRTC device.

**[0020]** The first WebRTC device or the connection management server may be configured to generate, in response to the receipt of the acknowledgement from the server, the identifier representing the first WebRTC device, the identifier representing the first WebRTC device being the second connection request specific.

**[0021]** The connection management server may be configured to associate the second connection request from the first WebRTC device with the inquiry to the connection management server from the second WebRTC device on a basis of a link address in the identifier representing the first WebRTC device obtained from the database.

**[0022]** Moreover, the communication system may be configured to establish the second connection with web real-time communication technology.

**[0023]** Still further, the communication system may be configured to carry, over the second connection, at least one of the following: real-time video data, video data file, document data, chat application data.

**[0024]** The expression "a number of" refers herein to any positive integer starting from one, e.g. to one, two, or three.

**[0025]** The expression "a plurality of" refers herein to any positive integer starting from two, e.g. to two, three, or four.

**[0026]** Various exemplifying and non-limiting embodiments of the invention both as to constructions and to methods of operation, together with additional objects and advantages thereof, will be best understood from the following description of specific exemplifying and non-limiting embodiments when read in connection with the accompanying drawings.

**[0027]** The verbs "to comprise" and "to include" are used in this document as open limitations that neither exclude nor require the existence of unrecited features. The features recited in dependent claims are mutually freely combinable unless otherwise explicitly stated. Furthermore, it is to be understood that the use of "a" or "an", i.e. a singular form, throughout this document does not exclude a plurality.

## BRIEF DESCRIPTION OF FIGURES

**[0028]** The embodiments of the invention are illustrated by way of example, and not by way of limitation, in the figures of the accompanying drawings.

Figure 1 illustrates schematically a communication environment in which the invention may be applied to.

Figure 2 illustrates schematically an example of a method according to an embodiment of the invention.

Figure 3 illustrates schematically aspects relating to a voice call connection establishment according to an embodiment of the invention.

Figure 4 illustrates schematically further aspects relating to an embodiment of the invention.

Figure 5 illustrates schematically an example of a communication device according to an embodiment of the invention.

Figure 6 illustrates schematically an example of a network node according to an embodiment of the invention.

## DESCRIPTION OF THE EXEMPLIFYING EMBODIMENTS

**[0029]** The specific examples provided in the description given below should not be construed as limiting the scope and/or the applicability of the appended claims. Lists and groups of examples provided in the description given below are not exhaustive unless otherwise explicitly stated.

**[0030]** Figure 1 illustrates schematically a communication environment into which a solution according to the present invention may be implemented to. A communication system forming the communication environment may comprise a first communication device 105 and a second communication device 110 both being configured to implement so called web real-time communication (WebRTC) technology among any other communication technology. The first and the second WebRTC devices 105, 110 may be any devices which are suitable for communicating with the manner as will be described. The devices 105, 110 may be configured to manage at least two connections established at least in part concurrently, wherein one of the connections may be established over Internet Protocol (IP) based network 130, such as

Internet, and the other connection may be established, at least in part, over a public switched telephone network (PSTN) 140. The first connection may be configured to carry other type data content that the other connection. For example, the first connection may carry voice in a form of a voice call, whereas the other connection, i.e. the second connection, may carry another type of data, such as real-time video data, video data file, document data, chat application data, or similar, over WebRTC technology. In some application area the first WebRTC device 105 is a client device suitable for establishing the connections to the second WebRTC device 110 representing a customer service center. At least one of the devices 105, 110 may be equipped with a camera for capturing image data conveyable as the video data content to the other party of the communication.

**[0031]** Further, the communication system may comprise a server 100 providing a service for initiating an establishment of a communication session between the parties, i.e. the first WebRTC device 105 and the second WebRTC device 110. Moreover, the communication system may comprise a WebRTC gateway device 115 for establishing the first connection for a voice call to the second WebRTC device 110, wherein the WebRTC gateway device 115 may be configured to transfer the call connection request to PSTN 140, such as a mobile communication network, from the IP based communication network 130. The PSTN 140 advantageously comprises at least one system for generating a data record for storing charging data associated to the communication session established. The charging data to be stored in the data record may be generated in accordance with voice call charging in the PSTN 140 by utilizing existing systems, and functionalities, in the PSTN 140. For example, the charging data may be generated, at least in part, by so called IP multimedia sub-system or by an intelligent network being involved in call connection establishment. Still further, the communication system may comprise a connection management server 120 for establishing the second connection between the parties, the second connection may be associated with the first connection for the voice call for establishing the communication session.

**[0032]** Next the invention will be described by referring to Figure 2 illustrating an example of a method according to the present invention. The example of the method is described, for sake of clarity, in a non-limiting scenario wherein the first connection is related to a voice call connection and the second connection is related to a real-time video connection. An establishment of a communication session may be initiated 210 in some manner. For example, a user of a first WebRTC device 105 may perform an action with the first WebRTC device 105 which causes signaling to a server 100. The action may refer to a trigger action generated with an input/output device of the first WebRTC device 105, for instance. Depending of an implementation the trigger action may e.g. be generated with an application executed by the first WebRTC device 105. The application may e.g. provide an interaction button to the user so that when the user activates the button, implemented e.g. with a touch screen of the first WebRTC device 105, a request to the server 100 is generated. The request for initiating the communication session may carry information identifying the first WebRTC device 105 and possibly information identifying the second WebRTC device 110 with whom the communication session is to be established. Moreover, the initiation of the communication session may be implemented so that the user willing to establish the communication session is required to be authenticated. The implementation in such as case may be such that the application through which the user initiates the session requests user credentials, such as a user name and a password, from the user for requesting the establishment of the communication session. Alternatively or in addition, the first WebRTC device 105 may comprise means for retrieving user specific data from a medium for authenticating the user. The means may e.g. be a reader device for the medium, such as for a smart card or for a tag or anything similar.

**[0033]** The server 100 receiving the request may be configured to perform one or more procedures for determining if the first WebRTC device 105 may be provided access to initiate the communication session. The procedure(s) may e.g. comprise identifying the user by comparing the received data to a stored data accessible to the server. For example, it may compare if the user credentials are correct or not. In case the server 100 determines that the communication session may be established with the first WebRTC device 105 it may be configured to acknowledge this to the first WebRTC device 105. In case the server 100 determines, for any reason, that the communication session may not be established with the first WebRTC device 105 it may be configured to cancel the initiation of the communication session 210 e.g. by generating a signal to the first WebRTC device 105 canceling the procedure.

**[0034]** The description of the method as schematically illustrated in Figure 1 will now be continued in a situation that the outcome of the determination by the server 100 is that the first WebRTC device 105 may establish the communication session. Otherwise, the establishment of the communication session may be canceled. Hence, in response to a receipt of an acknowledgement from the server 100 the first WebRTC device 105 may be configured to generate a first connection request for a voice call 215 to the second WebRTC device 110. The generation of the first connection request for the voice call 215 may be arranged so that the first WebRTC device 105 automatically, in response to the receipt of the acknowledgement, initiates the generation. Alternatively, it may be arranged so that the first WebRTC device 105 may be configured to prompt the user to initiate the generation of the first connection request for the voice call 215 in response to the receipt of the acknowledgement. The first connection request may comprise at least a network address of the first WebRTC device 105, or any similar identifier, by means of which the subscriber represented by the first WebRTC device 105 may be identified in a communication network, such as in a mobile communication network. For example, the identifier may be a subscriber number in the communication network, such as a MSISDN. The generation of the first connection request for the voice call 215 may be conveyed to a WebRTC gateway device 115, which may be configured to transfer the first connection request for the voice call to the PSTN 140 from the IP based communication network 130. Moreover, the first WebRTC device 105 may be configured to, in response to a receipt of an acknowledgement from the server 100, generate a second connection request 220 to a connection management server 120. The generation of the first connection request for the voice call 215 and the generation of the second connection request 220 may be arranged to occur concurrently at least in part or consecutively to each other. The second connection request generated to the connection management server 120 may comprise an identifier representing the first WebRTC device 105 towards a connection management server 120 implementing WebRTC based communication. The implementation of the WebRTC based communication may e.g. refer to an implementation that the connection management server 120 implements the WebRTC communication technology for providing data over the WebRTC technology between the parties of communication or the connection management server 120 may be involved in establishing the communication with WebRTC technology in any manner. Further, the identifier specific to the second connection may e.g. identify a communication environment, such as a unique video room name specific in a context of real-time video data to the subscriber in question. The identifier specific to the second connection may be generated by the first WebRTC device 105, by an application executed by the device 105, in response to the acknowledgement in step 210. Alternatively or in addition, the generation of the identifier specific to the second connection may be generated by any other entity, such as the connection management server 120, in response to an applicable signaling. The generated identifier specific to the second connection may also be delivered, i.e. stored, to data storage accessible to the second WebRTC device 110. In response to the receipt of the second connection request the connection management server 120 may be configured to store an indication on the received second connection request in data storage. The indication may comprise an identifier of the first WebRTC device 105, which identifies the first WebRTC device 105 either directly or indirectly. An example of an identifier identifying the first WebRTC device 105 directly may e.g. be a network address of the first WebRTC device 105 whereas the identifier specific to the second connection may be an example of an identifier indirectly identifying the first WebRTC device 105. In some embodiment the identifier may comprise a link address for establishing the second connection. The link address may comprise a network address and/or a virtual address identifying the video session to be established.

**[0035]** Next, in response to an establishment of the first connection for the voice call 230 experienced by the second WebRTC device 110 further operations may be conducted. The establishment of the first connection for voice call 230 shall in this context be understood at least to cover the following: a receipt of the first connection request in the second WebRTC device 110, accepting the first connection request by the second WebRTC device 110 for establishing the voice call connection over the first connection.

**[0036]** In response to the establishment of the voice call connection over the first connection 230 mentioned above the second WebRTC device 110 may be configured to obtain an identifier of the first WebRTC device 110 from the first connection request for the voice call. Next, the second WebRTC device 110 may be configured to generate an inquiry to the connection management server 120, to any other entity, such as to a predetermined data storage, for determining 235 if a connection request is received from the first WebRTC device 105 with which the first connection for the voice call is established by the second WebRTC device 110. The inquiry to the connection management server 120 may carry an identifier representing the first WebRTC device 105 from which the voice call connection over the first connection is established. The connection management server 120 may be configured to respond to the inquiry in a manner providing information if the connection management server 120, or the data storage, stores in indication that the connection request is received. In case there is no indication on the connection request the second WebRTC device 110 may be configured to terminate the establishment of the communication session. Alternatively, in response to a determination that the connection management server 120 stores an indication on the connection request from the first WebRTC device 105 a communication session may be established 240 between the first WebRTC device 105 and the second WebRTC device 110. As a result, the communication session established in the manner as described comprises both the voice call connection over the first connection and the second connection so that the voice content may be delivered, at least in part, through another route than another content. The other content may e.g. be a real-time video data, data files carrying video data or document data, chat application data, or any combination of these, which other content is carried by utilizing WebRTC technology at least in part. The connection management server 120 may advantageously be configured to implement one or more functionalities which enable carrying the data between the parties over the second connection.

**[0037]** As indicated the connection management server 120 may be configured to convey the data over the second connection. In a further embodiment the transfer of data over the second connection may be implemented so that the connection management server 120 may be configured to link the first and the second WebRTC devices 105, 110 together, but the data transfer is performed over so-called peer-to-peer connection directly between the communicating parties. This may e.g. be achieved so that the connection management server 120 transmits information on at least a network address, such as an IP address, of the other communicating party to each communicating party i.e. the network address of the first WebRTC device 105 to the second WebRTC device 110 and vice versa. In response to a receipt of the network address, or addresses, the second connection may be established directly between the parties as a peer-to-peer connection, for example. In this kind of an embodiment a role of the connection management server 120 is, more or less, to operate as an entity, which enables connection between the devices 105, 110 at least in part by means of identifying that the communicating parties are searching each other.

**[0038]** In an embodiment of the invention in which the second WebRTC device 105 is configured to inquire from a data storage, such as from database, if an indication on the connection request is stored, it may be arranged that the response to the inquiry comprises a link address to the connection in question. The link address may e.g. direct, by activating the link address, the second WebRTC device to initiate the connection towards the connection management server 120 wherein the connection e.g. carries an identifier of the connection in question. In this manner the connection management server 120 may associate the connection requests received from the first WebRTC device 105 and from the second WebRTC device 110 together. The inquiry performed by the second WebRTC device 110 to the data storage may be arranged so that the inquiry is performed directly to the data storage or indirectly over another entity. The other entity may e.g. be a back-end device or a server device, such as a web server, which may be instructed to access the data storage by the second WebRTC device 110 with signaling, for instance.

**[0039]** Next, some aspects with respect to an establishment of the first connection for the voice call, as described above, is discussed by referring to Figure 3. As already mentioned the WebRTC gateway device 115 may be configured to transfer the call connection request over the first connection to PSTN 140. The first connection may be initiated by the user of the first WebRTC device 105 with an application software executed by the first WebRTC device 105. The application software may e.g. be an internet browser which supports WebRTC technology. In response to a receipt of the first connection request for the voice call in the WebRTC gateway device 115 the WebRTC gateway device 115 may be configured to obtain an identifier, such as a network address, representing the first WebRTC device 105 and to generate a request towards one or more network elements implementing network functionalities in the PSTN side. According to an embodiment the WebRTC gateway device 115 may be configured to generate the request to a IP multimedia subsystem 310 with an applicable protocol, such as with a Session Initiation Protocol, SIP. As mentioned the request advantageously carries an identifier for the first WebRTC device 105 and the IP multimedia subsystem may obtain user related information from a register storing subscriber data, such as HSS (Home Subscriber Server). In response to operations performed by the IP multimedia subsystem 310 the voice call connection request is forwarded, as a SIP message, to a network element 320 configured to perform at least so-called SIP trunk functionalities. The SIP trunk network element 320 may be configured to facilitate the voice over IP connection to a network element 330 configured to perform PSTN related operations, such as a generation of charging data. The network element 330 may e.g. refer to an Intelligent Network (IN) by means of which dedicated charging, such as company specific pricing, may be applied to so that the receiver, i.e. the entity represented by the second WebRTC device 110 pays costs of the communication session. In other words, the SIP trunk network element performs necessary manipulation to at least some data carried in the SIP message, such as any number manipulation with respect to subscription represented by the first WebRTC device 105 or the second WebRTC device 110 or both. Further, the SIP trunk network element may deliver the first connection request to the second WebRTC device 110, which is configured to execute an application suitable to managing a call connection over PSTN. The identifier of the calling party and the called party may be any applicable network identifier, such as a MSISDN or Session Initiation Protocol (SIP) address.

**[0040]** It is worthwhile to mention that network entities for delivering the first connection for the voice call as described above are non-limiting examples of suitable entities. However, the first connection may be arranged in any other manner over the PSTN than described. For example, in some implementation the IP Multimedia Subsystem may be omitted, and the SIP trunk network element 320 may be configured to take care of the establishment of the first connection for the voice call.

**[0041]** Figure 4 illustrates schematically further aspects of the present invention. As mentioned the established communication session comprises both the first connection for the voice call and the second connection for other data so that the voice data may be delivered, at least in part, through another route than the other data. The first WebRTC device 105 may be configured to initiate an establishment of the communication session in the manner as described. The initiation may be performed by an application executed by the first WebRTC device 105 wherein the application supports the WebRTC technology. A non-limiting example of the application may e.g. be an Internet browser, or a dedicated communication application, configured to support the WebRTC technology. Since the communication initiated by the first WebRTC device 105 is divided to two separate communication paths, as discussed, the second WebRTC device 110 may be arranged to execute such an application or applications which is/are suitable for control the communication from each of the communication paths. According to an embodiment the second WebRTC device 110 may be configured to execute two applications 420, 430. The first application 420 may be configured to support an establishment of the second connection, and, thus, the other data content delivered over the second connection, such as video data, whereas the second application 430 may be configured to support the first connection for the voice call received at least in part over the PSTN 140. Again, the first application 420 executed by the second WebRTC device 110 may be an Internet browser, or a dedicated communication application, configured to support the WebRTC technology. The second application 430, in turn, may e.g. be voice call application providing a signaling interface towards a communication network delivering the voice call over the first connection between the communicating parties. Advantageously, the delivery of data through the communication paths is synchronized so that a timing of the voice carried over the voice call connection and the other data, such as image data or chat data, carried over the second connection are in sync at the receiver end. This may e.g. be achieved by including time stamps in the delivered data at the transmitter end, wherein the output of the pieces of data is performed based on the time stamps.

**[0042]** The description of an embodiment of the invention above is mainly based on an idea that the first WebRTC device 105 is configured to establish the voice call connection over the first connection and the second connection towards the mentioned entities, such as the WebRTC gateway device 115 and the connection management server 120. However, the present invention is not limited only to such an implementation, but it may be arranged that the first WebRTC device 105 initiates all communication i.e. both the first connection for the voice call and the second connection to a same network node, which is configured to divide the first connection and the second connection towards different entities, such as towards the WebRTC gateway device 115 and the connection management server 120. The network node for dividing the connections may a distinct network node to the mentioned ones or it may even be one of the following: a server 100, the WebRTC gateway device 115 or the connection server 120.

**[0043]** As mentioned the communication system forming the communication environment may comprise a first communication device 105 and a second communication device 110 both being configured to implement so called web real-time communication (WebRTC) technology among any other communication technology. Moreover, the communication system may comprise a server 100, a WebRTC gateway device 115 and a connection management server 120 for establishing a communication session comprising both the first connection for the voice call and the second connection. An example of the first or the second WebRTC device 105, 110 is schematically illustrated in Figure 5. The WebRTC device 105, 110 may be configured to implement at least part of the method for establishing the communication session as described. The execution of the method, or at least some portion of it, may be achieved by arranging a processor 510 to execute at least some portion of computer program code 521a-521 n stored in a memory 520 causing the processor 510, and, thus, the WebRTC device 105, 110 to implement one or more method steps as described. The computer program code 521a-521 n, when executed by the processor, may implement at least some of the functionalities of the application(s) 410 or 420, 430, as referred in the context of the description of Figure 4. In other words, the processor 510 may be arranged to access the memory 520 and to retrieve and to store any information therefrom and thereto. Moreover, the processor 510 may be configured to control the communication through one or more communication interface 530, such as through a plurality of Local Area Network (LAN) interfaces. Hence, the communication interface 530 may be arranged to implement, possibly under control of the processor 510, a corresponding communication protocol, such as an IP, in question. Further, the WebRTC device 105, 110 in question may comprise one or more input/output devices 540 for inputting and outputting information. Such input/output devices may e.g. be keyboard, buttons, touch screen, display, loudspeaker, camera and so on. In some implementation of the WebRTC device 105, 110 at least some of the in-put/output devices may be external to the WebRTC device 105, 110 and coupled to it either wirelessly or in a wired manner. For sake of clarity, the processor 510 herein refers to any unit or a plurality of units suitable for processing information and control the operation of the WebRTC device 105, 110 in general at least in part, among other tasks. The mentioned operations may e.g. be implemented with a microcontroller solution with embedded software. Similarly, the invention is not limited to a certain type of memory 520 only, but any memory unit or a plurality of memory units suitable for storing the described pieces of information, such as portions of computer program code and/or parameters, may be applied in the context of the present invention. Even if it is disclosed that the method may be implemented with one WebRTC device 105, 110, it may also be arranged that the implementation of the method, or at least some portions of it, is performed in multiple devices operatively coupled to each other either directly or indirectly as a distributed implementation. In case the WebRTC device 105, 110 is integrated with to another network entity the functionality of the WebRTC device 105, 110 is advantageously implemented in the manner as described.

**[0044]** Figure 6, in turn, illustrates schematically an example of a network node, such as a server 100, a WebRTC gateway device 115, or a connection management server 120. All these devices may comprise a processor 610, a memory 620 storing data, such as portions of computer program code 621a-621n, a communication interface 630 and possibly input/output devices 640 for inputting and outputting information. Depending on a task the device in question may be implemented so that the communication interface is configured to implement one or more communication technology, such as the WebRTC gateway device 115 may be configured to execute both the applied technology in the Internet side as well as the applied technology in the PSTN side and even implement a conversion of the communication between the two entities.

**[0045]** The description of at least some aspects of the present invention is performed in a communication system with separate network devices, such as the server 100, the WebRTC gateway device 115, the connection management server 120. Generally speaking, the communication system, according to an embodiment of the invention, may comprise means for performing the described method in the communication system. The means may be arranged in the same computing device, such as in a server, or it may be distributed computing environment, in which the operations and functionalities are shared between separate network nodes.

**[0046]** The specific examples provided in the description given above should not be construed as limiting the applicability and/or the interpretation of the appended claims. Lists and groups of examples provided in the description given above are not exhaustive unless otherwise explicitly stated.

# REFERENCES CITED IN THE DESCRIPTION

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## PATENTKRAV

1. Fremgangsmåde til etablering af en kommunikationssession, hvilken fremgangsmåde omfatter:

modtagelse, ved hjælp af en server (100), af en anmodning om etablering af en kommunikationssession fra en første net-sandtidskommunikations-, WebRTC-, indretning (105) til en anden net-sandtidskommunikations-, WebRTC-, indretning (110),
bekræftelse, ved hjælp af serveren (100), af anmodningen til den første WebRTC-indretning (105),
og som svar på en modtagelse af bekræftelsen:

generering, ved hjælp af den første WebRTC-indretning (105), af en første forbindelsesanmodning (215) om et taleopkald, hvilken første forbindelsesanmodning omfatter i det mindste en netværksadresse for den første WebRTC-indretning (105) imod en WebRTC-gateway-indretning (115), for etablering af en første forbindelse til taleopkald (230) til den anden WebRTC-indretning (110), hvilken WebRTC-gateway-indretning (115) er konfigureret til at overføre den første forbindelsesanmodning til et offentligt telefonnet (140),
generering, ved hjælp af den første WebRTC-indretning (105), af en anden forbindelses-anmodning (220), som omfatter en identifikator, som repræsenterer den første WebRTC-indretning (105) imod en forbindelseshåndteringsserver (120), som implementerer WebRTC-baseret kommunikation,

og som svar på en etablering af den første forbindelse for taleopkald (230) af den anden WebRTC-indretning (110):

generering af en forespørgsel til forbindelseshåndteringsserveren (120) af den anden WebRTC-indretning (110) for bestemmelse (235) om en forbindelsesanmodning er modtaget fra den første WebRTC-indretning (105), hvormed den første forbindelse for taleopkaldet etableres af den anden WebRTC-indretning (110), hvilken forespørgsel til forbindelseshåndteringsserveren (120) bærer en identifikator, som repræsenterer den første WebRTC-indretning (105), fra hvilken den første forbindelse for taleopkaldet er etableret, og som svar på en bestemmelse (235), af den anden WebRTC-indretning (105), at forbindelseshåndteringsserveren (120) lagrer en indikation om forbindelses-anmodningen fra den første WebRTC-indretning (105), etablering (240) af en kommunikationssession imellem den første WebRTC-indretning (105) og den anden WebRTC-indretning (110), hvilken kommunikationssession omfatter både den første forbindelse for taleopkaldet og en anden forbindelse.

2. Fremgangsmåde ifølge krav 1, hvor anmodningen om etablering af en kommunikationssession fra en første WebRTC-indretning (105), omfatter data for autentificering af den første WebRTC-indretning (105) af serveren (100).

3. Fremgangsmåde ifølge ethvert af de foregående krav, hvor generering, af en af følgende: et IP-multimedie sub-system, som foreligger i det offentlige telefonnetværk (140), et intelligent-netværk, som er involveret i opkaldsforbindelsesetablering og som foreligger i det offentlige telefonnetværk (140), af dataregistrering til lagring af takseringsdata knyttet til kommunikationssessionen, som er etableret, hvor takseringsdataene genereres i overensstemmelse med taleopkaldstaksering.

4. Fremgangsmåde ifølge ethvert af de foregående krav, hvor identifikatoren, som repræsenterer den første WebRTC-indretning (105), fra hvilken den første forbindelse for taleopkaldet etableres, opnås af den anden WebRTC-indretning (110) ved at forespørge om denne fra en database, som lagrer identifikatoren knyttet til netværksadressen for den første WebRTC-indretning (105), hvilken forespørgsel til databasen omfatter en netværksadresse for den første WebRTC-indretning (105), som er opnået fra den første forbindelsesanmodning for taleopkaldet modtaget af den anden WebRTC-indretning (110).

5. Fremgangsmåde ifølge ethvert af de foregående krav, hvor identifikatoren, som repræsenterer den første WebRTC-indretning (105), er den anden forbindelse specifik og genereret som svar på modtagelsen af bekræftelsen fra serveren (1) af den første WebRTC-indretning (105).

6. Fremgangsmåde ifølge ethvert af de foregående krav, hvor identifikatoren, som repræsenterer den første WebRTC-indretning (105), omfatter en link-adresse til forbindelseshåndteringsserveren (120) til at associere, af forbindelseshåndteringsserveren (120), den anden forbindelsesanmodning fra den første WebRTC-indretning (105) med

forespørgslen til forbindelseshåndteringsserveren (120) fra den anden WebRTC-indretning (110).

7. Fremgangsmåde ifølge ethvert af de foregående krav, hvor den anden forbindelse etableres med WebRTC-sandtidskommunikationsteknologi.

8. Fremgangsmåde ifølge ethvert af de foregående krav, hvor data føres over den anden forbindelse, idet dataene er i det mindste én af følgende: sand-tids-videodata, videodatafil, dokumentdata, chat-applikationsdata.

9. Kommunikationssystem til etablering af en kommunikationssession, hvilket system omfatter:

en første net-sandtids-kommunikations-, WebRTC-, indretning (105) og en anden net-sandtids-kommunikations-, WebRTC-, indretning (110),
en server (100), som er konfigureret til at bekræfte, til den første WebRTC-indretning (105), en anmodning om etablering af en kommunikationssession fra den første WebRTC-indretning (105) til den anden WebRTC-indretning (110),
en WebRTC-gateway-indretning (115), som er konfigureret til, som svar på en modtagelse af en første forbindelsesanmodning fra den første WebRTC-indretning (105) om et taleopkald, omfattende i det mindste en netværksadresse for den første WebRTC-indretning (105), at etablere en første forbindelse for taleopkaldet (230) til den anden WebRTC-indretning (110), hvilken WebRTC-gateway-indretning (115) er konfigureret til at overføre den første forbindelsesanmodning til et offentligt telefonnetværk (140),
en forbindelseshåndteringsserver (120), som er konfigureret til at modtage en anden forbindelsesanmodning fra den første WebRTC-indretning (105), hvilken anden forbindelsesanmodning omfatter en identifikator, som repræsenterer den første WebRTC-indretning (105), hvilken forbindelseshåndteringsserver (120) er konfigureret til at implementere WebRTC-baseret kommunikation,
hvor, som svar på en etablering af den første forbindelse for taleopkald (230) af den anden WebRTC-indretning (110), systemet er konfigureret således at:

den anden WebRTC-indretning (110) er konfigureret til at generere, som svar på etablering af den første forbindelse for taleopkaldet (230), en forespørgsel til forbindelseshåndteringsserveren (120) for bestemmelse (235) om en forbindelsesanmodning er modtaget fra den første WebRTC-indretning (105), hvormed den første forbindelse for taleopkaldet er etableret af den anden WebRTC-indretning (110), hvilken forespørgsel til forbindelseshåndterings-serveren (120) er konfigureret til at føre en identifikator, som repræsenterer den første WebRTC-indretning (105), hvorfra den første forbindelse for tale-opkaldet er etableret,

og hvor, som svar på en bestemmelse (235) af den anden WebRTC-ind-retning (110), at forbindelseshåndteringsserveren (120) lagrer en indikation om forbindelsesanmodningen fra den første WebRTC-indretning (105), er systemet indrettet til at etablere (240) en kommunikationssession imellem den første WebRTC-indretning (105) og den anden WebRTC-indretning (110), hvilken kommunikationssession omfatter både den første forbindelse for taleopkaldet og en anden forbindelse.

10. Kommunikationssystem ifølge krav 9, hvor serveren (110) er konfigureret til at bekræfte den første WebRTC-indretning (105) om anmodningen for etablering af en kommunikationssession fra den første WebRTC-indretning (105) til den anden WebRTC-indretning (110), som svar på en autentifikation af den første WebRTC-ind-retning (105) af serveren (100) baseret på autentifikationsdata modtaget i anmodningen.

11. Kommunikationssystem ifølge ethvert af kravene 9 eller 10, hvor én af følgende: et IP-multimedie subsystem, som foreligger i det offentlige telefonnetværk (140), et intelligent netværk, som er involveret i opkaldsforbindelsesetablering og foreligger i det offentlige telefonnetværk (140), er konfigureret til at generere data registrering for lag-ring af takseringsdata knyttet til kommunikationssessionen, som er etableret, hvor takseringsdataene genereres i overensstemmelse med taleopkaldstaksering.

12. Kommunikationssystem ifølge ethvert af de foregående krav 9-11, hvor den anden WebRTC-indretning (110) er konfigureret til at opnå identifikatoren, som repræsenterer den første WebRTC-indretning (105) hvorfra den første forbindelse for taleopkaldet er etableret, ved at forespørge denne fra en database, som lagrer identifikatoren, som er associeret med netværksadressen for den første WebRTC-indretning (105), hvilken forespørgsel til databasen omfatter en netværksadresse for den første WebRTC-indretning (105), som er opnået fra den første forbindelsesanmodning for taleopkaldet, som er modtaget af den anden WebRTC-indretning (110).

13. Kommunikationssystem ifølge ethvert af de foregående krav 9-12, hvor den første WebRTC-indretning (105) er konfigureret til at generere, som svar på modtagelsen af bekræftelsen fra serveren (100), identifikatoren, som repræsenterer den første WebRTC-indretning (105), hvilken identifikator, som repræsenterer den første WebRTC-indretning (105), er den anden forbindelse specifik.

14. Kommunikationssystem ifølge ethvert af de foregående krav 9-13, hvor forbindelseshåndteringsserveren (120) er konfigureret til at associere den anden forbindelsesanmodning fra den første WebRTC-indretningen (105) med forespørgslen til forbindelseshåndteringsserveren (120) fra den anden WebRTC-indretning (110), på basis af en link-adresse i identifikatoren, som repræsenterer den første WebRTC-indretning (105).

15. Kommunikationssystem ifølge ethvert af de foregående krav 9-14, hvilke kommunikationssystem er konfigureret til at etablere den anden forbindelse med net-sand-tidskommunikationsteknologi.

16. Kommunikationssystem ifølge ethvert af de foregående krav 9-15, hvor kommunikationssystemet er konfigureret til at føre, via den anden forbindelse, data, hvilke data er i det mindste én af følgende: sand-tidsvideodata, videodatafil, dokumentdata, chat-applikationsdata.

# DRAWINGS

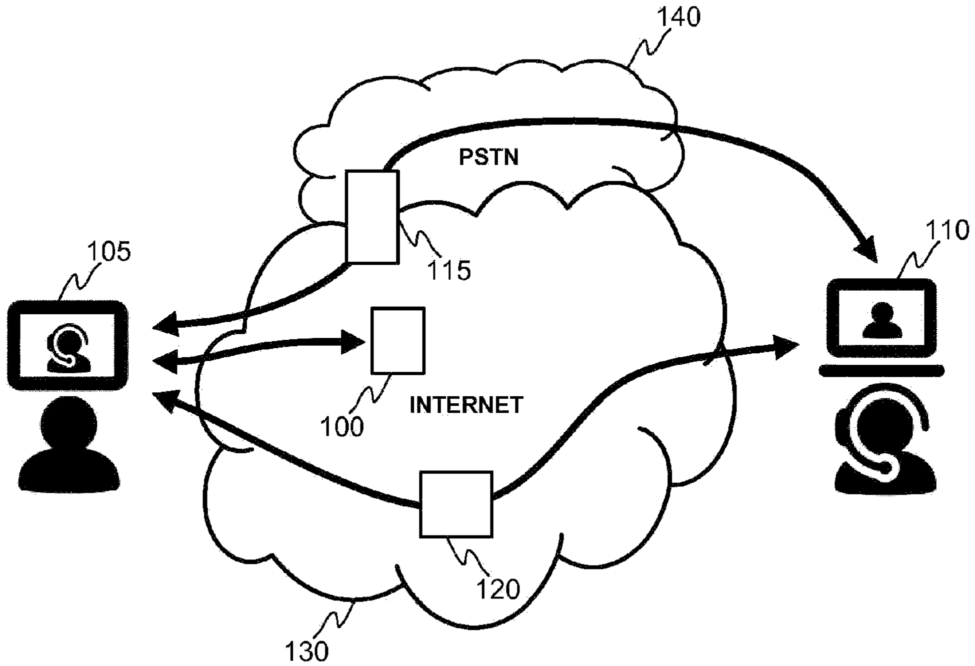


FIGURE 1

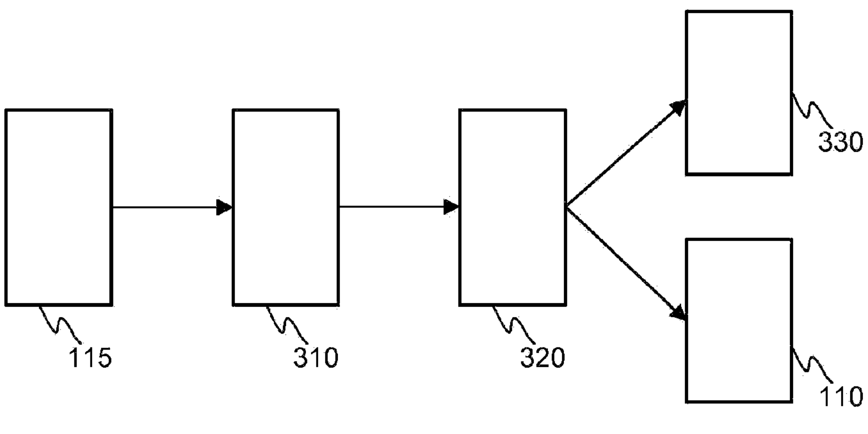


FIGURE 3

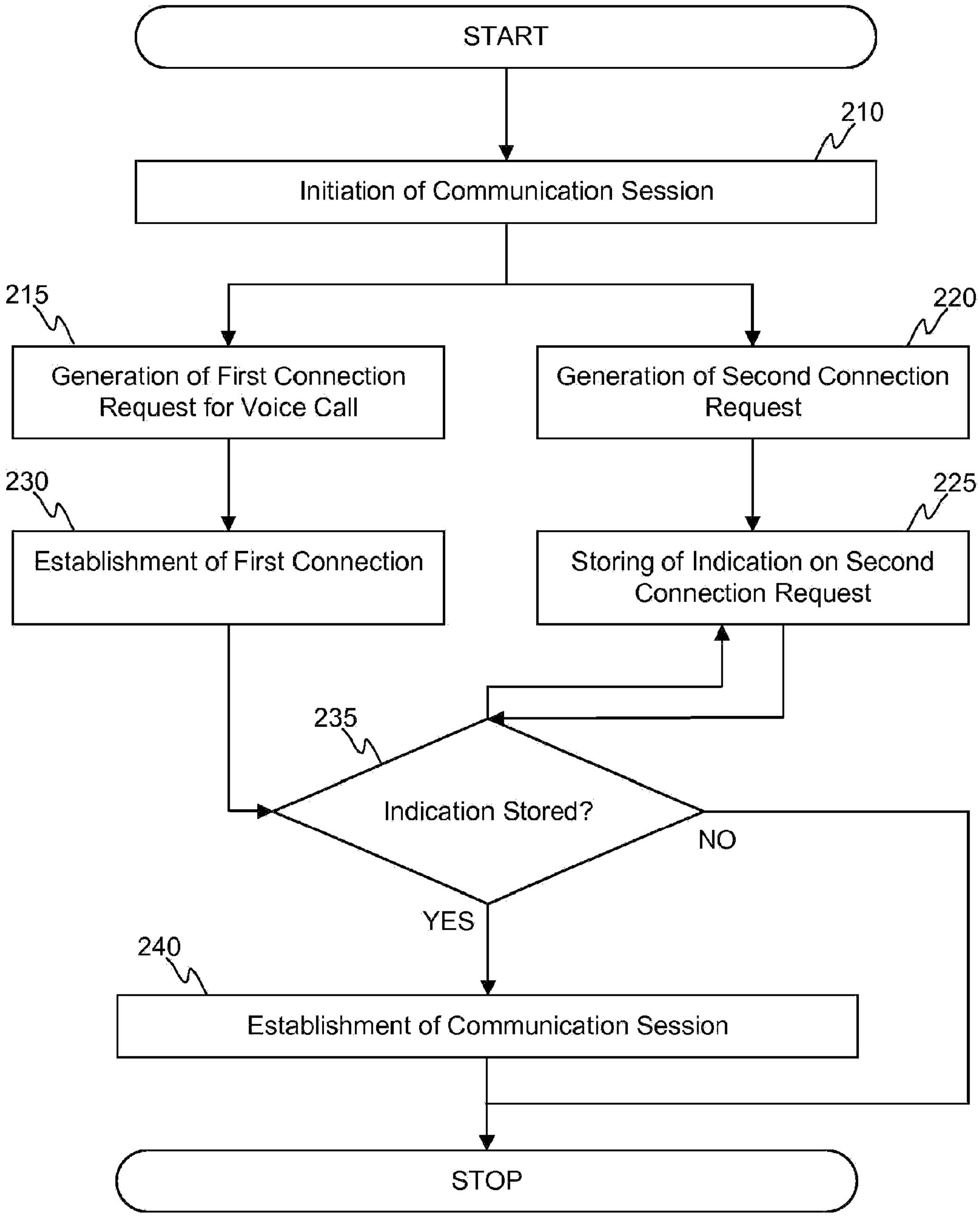
START
210
Initiation of Communication Session
215
Generation of First Connection Request for Voice Call
220
Generation of Second Connection Request
230
Establishment of First Connection
225
Storing of Indication on Second Connection Request
235
Indication Stored?
NO
YES
240
Establishment of Communication Session
STOP

FIGURE 2

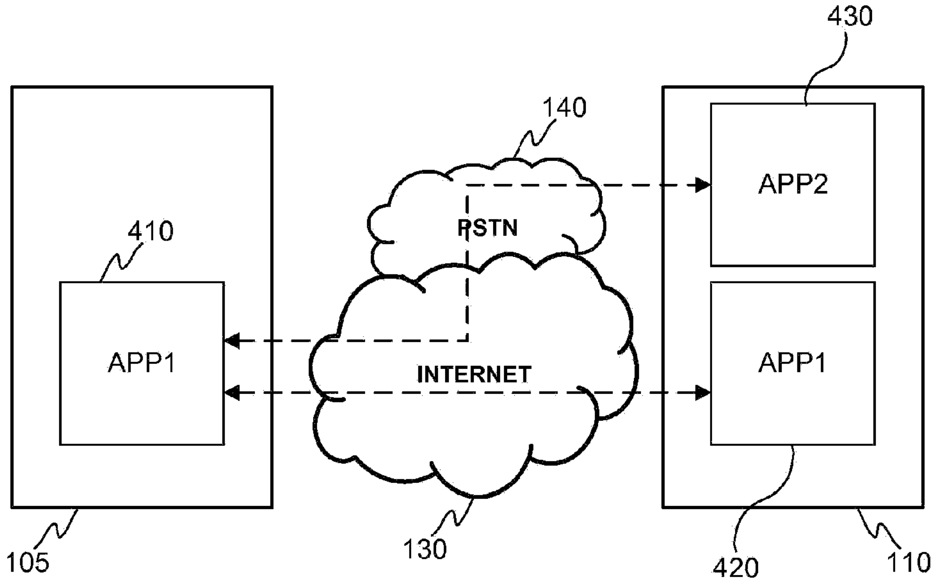


**FIGURE 4**

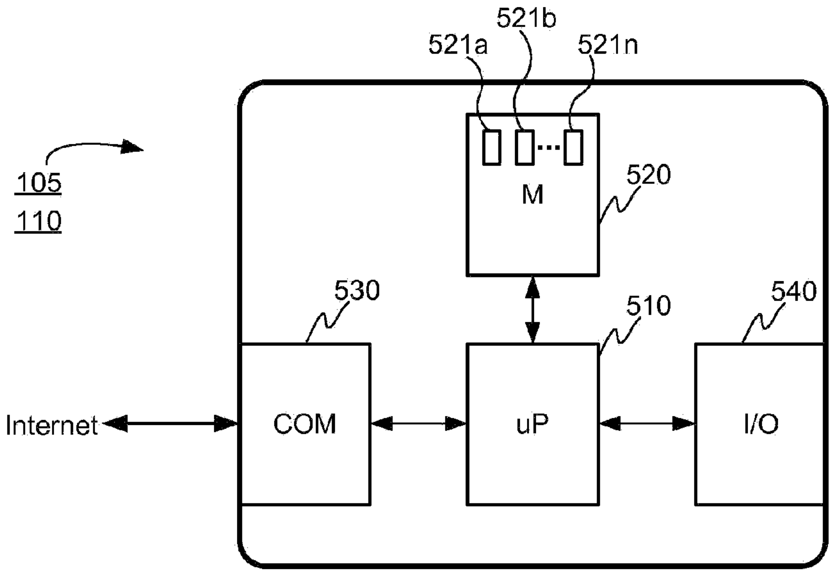


**FIGURE 5**

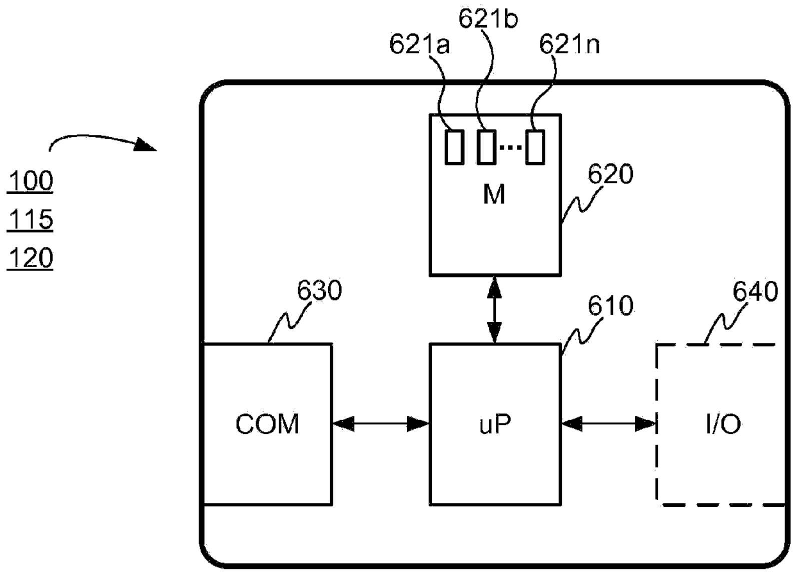


**FIGURE 6**