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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
12/071,005	02/14/2008	Louis Holder	Y2108-00123 DIV1	7617

7590 09/21/2016

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ART UNIT	PAPER NUMBER
2467	

MAIL DATE	DELIVERY MODE
09/21/2016	PAPER

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UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

Ex parte LOUIS HOLDER and JEFFREY CITRON

Appeal 2015-006713
Application 12/071,005
Technology Center 2400

Before CAROLYN D. THOMAS, ERIC B. CHEN, and
JOSEPH P. LENTIVECH, *Administrative Patent Judges*.

CHEN, *Administrative Patent Judge*.

DECISION ON APPEAL

This is an appeal under 35 U.S.C. § 134(a) from the final rejection of claims 8–46. Claims 1–7 have been cancelled. (*See* App. Br. 14.) We have jurisdiction under 35 U.S.C. § 6(b). We affirm-in-part.

STATEMENT OF THE CASE

Appellants' invention relates to Internet telephony that uses of Session Initiated Protocol technology. (Spec. 4:11–12.)

Claims 8 and 11 are exemplary, with disputed limitation in italics:

> 8. An internet telephony system configured to use Session Initiation Protocol (SIP) signaling to setup a communication of streaming packets, the internet telephony system comprising:
>
> a relay configured to relay streaming packets of the communication between a caller and a call destination;
>
> a server configured to receive, process and transmit SIP signaling messages to setup the communication between the caller and the call destination and to select the relay to use for the communication, *the selection being based at least on the quality of the communication.*
>
> 11. The internet telephony system of claim 8, *wherein the server is configured to select the relay to decrease the latency of the communication.*

Claims 8–10, 12–14, 16–22, 24–26, 28–34, 36–38, and 40–43 stand rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien (US 7,369,535 B2; May 6, 2008) and Kobayashi (US 2005/0246347 A1; Nov. 3, 2005).

Claims 15, 27, and 39 stand rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien, Kobayashi, and Sylvain (US 2004/0120498 A1; June 24, 2004).

Claims 11, 23, and 35 stand rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien, Kobayashi, and Mangal (US 6,865,398 B2; Mar. 8, 2005).

Claims 44–46 stand rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien, Kobayashi, and to Kleyman (US 2004/0244010 A1; Dec. 2, 2004).[1]

ANALYSIS

§ 103 Rejection—O'Brien and Kobayashi

We are unpersuaded by Appellants' arguments (App. Br. 4–9; *see also* Reply Br. 1–4) that the combination of O'Brien and Kobayashi would not have rendered obvious independent claim 8, which includes the limitation "the selection being based at least on the quality of the communication."

The Examiner found that the VoIP packets of O'Brien correspond to the limitation "streaming packets." (Final Act. 4.) The Examiner further found that the relay control section of Kobayashi corresponds to the limitation "the selection being based at least on the quality of the communication." (*Id.* at 5; *see also* Ans. 8.) In particular, the Examiner found that "the lower the capacity/load gets, the worse the quality (communication quality) gets." (Ans. 7.) The Examiner concluded that "it would be obvious . . . to implement selecting relays based on QoS as taught by Kobayashi with . . . O'Brien." (*Id.* at 10.) We agree with the Examiner.

[1] Appellants do not present any separate arguments with respect to the rejections of dependent claims 44–46 under 35 U.S.C. § 103(a). Thus, any such arguments are deemed to be waived.

O'Brien relates to voice-over-IP telephony. (Abstract.) O'Brien explains that Voice over Internet Protocol (VoIP) Real Time Protocol (RTP) includes "forcing packets carrying media in a VoIP call through network elements." (Col. 1, ll. 28–31.) O'Brien explains that "[t]he quality of a VoIP call is insured by controlling the path of the media stream to ensure that these voice packets traverse a known . . . IP network," such that "[t]he path of the voice packets can avoid congested networks." (Col. 2, ll. 26–29.)

Kobayashi relates to "a cache server and a network system having cache servers." (¶ 3.) Kobayashi explains that "a relay control section selects a relay server that is necessary for setting a path suitable for carrying out an automatic cache updating operation, a link prefetching operation, and a cache server cooperating operation, based on QoS path information." (¶ 37.) In reference to Figure 9, Kobayashi provides an example in which relay control section 110 uses "relay servers M**301** and M**302** . . . to check whether it is possible or not to set a path from the Web server S**1** to the QoS path reference relay control cache server C**301** without passing through the congestion section [link L1]." (¶ 131.) Because Kobayashi explains that path selection is based on the avoidance of congestion, Kobayashi teaches or suggests the limitation "the selection being based at least on the quality of the communication."

A person of ordinary skill in the art would have recognized that incorporating the relay control section of Kobayashi with the VoIP of O'Brien would improve O'Brien by providing a suitable uncongested path based on QoS. *See KSR Int'l Co. v. Teleflex Inc.*, 550 U.S. 398, 417 (2007) ("[I]f a technique has been used to improve one device, and a person of ordinary skill in the art would recognize that it would improve similar devices in the same way, using the technique is obvious unless its actual application is beyond his or her skill."). Thus, we agree with the Examiner (Ans. 10) that modifying O'Brien to incorporate the relay control section of Kobayashi would have been obvious.

Appellants argue that "the QoS path information [of Kobayashi] is not the same nor equivalent to the 'Quality of Communication' as claimed, specifically the QoS path information is described in Kobayashi as simply available bandwidth/link load." (App. Br. 4; *see also* Reply Br. 1–2.) However, O'Brien expressly states that "quality of a VoIP call is insured by controlling the path of the media stream to ensure that these voice packets traverse a known . . . IP network" and "[t]he path of the voice packets can avoid congested networks." (Col. 2, ll. 26–29.) Thus, the VoIP call quality of O'Brien is directly correlated to the QoS path information of Kobayashi for avoiding congested paths.

Appellants further argue that "[c]ache servers (as used in Kobayashi) for data by their very nature are an anathema to Voice" because "[a] cache server on the other hand CACHES information packets until they are ready to transport." (App. Br. 6 (emphasis omitted).) Accordingly, Appellants argue, "one of ordinary skill in the art of VoIP communication would not sacrifice quality of communication for a reduction of congestion which would render the VoIP communication of O'Brien unsatisfactory, and thus the Office's motivation to combine Kobayashi and O'Brien is without merit." (App. Br. 9 (emphasis omitted); *see also* Reply Br. 3–4.) However, the Examiner cited Kobayashi for the general teaching of using a relay control to select a suitable path based on QoS is well known, rather than bodily incorporating the cache server of Kobayashi into the VoIP of O'Brien. (*See* Ans. 8.)

Thus, we agree with the Examiner that the combination of O'Brien and Kobayashi would have rendered obvious independent claim 8, which includes the limitation "the selection being based at least on the quality of the communication."

Accordingly, we sustain the rejection of independent claim 8 under 35 U.S.C. § 103(a). Claims 9, 10, 12–14, and 16–19 depend from claim 8, and Appellants have not presented any additional substantive arguments with respect to these claims. Therefore, we sustain the rejection of claims 9, 10, 12–14, and 16–19 under 35 U.S.C. § 103(a), for the same reasons discussed with respect to independent claim 8.

Independent claims 20 and 32 recite limitations similar to those discussed with respect to independent claim 8, and Appellants have not presented any additional substantive arguments with respect to these claims. We sustain the rejection of claims 20 and 32, as well as dependent claims 21, 22, 24–26, 28–31, 33, 34, 36–38, and 40–43, for the same reasons discussed with respect to claim 8.

§ 103 Rejection—O'Brien, Kobayashi, and Sylvain

We are unpersuaded by Appellants' arguments (App. Br. 9–11*; see also* Reply Br. 4–5) that the combination of O'Brien, Kobayashi, and Sylvain would not have rendered obvious dependent claim 15, which includes the limitation "wherein the server is configured to select the relay based on a SIP Invite message."

The Examiner found that the SIP INVITE message of Sylvain corresponds to the limitation "wherein the server is configured to select the relay based on a SIP Invite message." (Ans. 12–13; *see also* Final Act. 7.) The Examiner concluded that "it would have been obvious . . . to implement select[ing] the relay to decrease the latency of the communication as taught by Sylvain with the combined teachings of and O'Brien, Jr. and Kobayashi for the purpose further managing the routing of streaming media between users." (Final Act. 7 (emphasis omitted).) We agree with the Examiner.

Sylvain relates to "associating multimedia services with traditional telephony services in an efficient manner." (¶ 1.) In one embodiment, Sylvain explains that "SIP call servers **26** . . . facilitate VoP communications via the packet network **22**," utilizing the known Session Initiation Protocol (SIP). (¶ 23.) Figure 5 of Sylvain illustrates an embodiment in which single service node 28 supports multimedia clients A and B. (¶ 42.) Sylvain explains "[a]ssuming that user A wishes to initiate a video session with user B, sufficient action is taken to trigger multimedia client A (**14**) to send a SIP INVITE message to multimedia client B (**14**)." (¶ 46.) Because Sylvain explains that SIP call servers 26 facilitate VoP communications and user A sends user B a SIP INVITE, Sylvain teaches or suggests the limitation "wherein the server is configured to select the relay based on a SIP Invite message."

A person of ordinary skill in the art would have recognized that incorporating the SIP INVITE message of Sylvain with the VoP of O'Brien, as modified by Kobayashi, would improve O'Brien by providing the ability to facilitate communications via Session Initiation Protocol (SIP). *See KSR*, 550 U.S. at 417. Thus, we agree with the Examiner (Final Act. 7) that modifying O'Brien and Kobayashi to incorporate the SIP INVITE message of Sylvain would have been obvious.

Appellants argue that "Sylvain does not disclose the use of the telephone switch to relay streaming packets" but instead, "[t]he telephony switch of Sylvain is a conventional switch used to communicate over the PSTN (Public Switched Telephone Network)." (App. Br. 10–11; *see also* Reply Br. 4–5.) However, the Examiner cited to O'Brien, rather than Sylvain for teaching the limitation "streaming packets." (Final Act. 4.)

Appellants further argue that "the motivation relied upon by the Office is related to reduction of latency which is in no manner linked to the proposed modification." (App. Br. 11.) However, the Examiner also provides the rationale the combination of O'Brien, Kobayashi, and Sylvain provides the advantage of "managing the routing of streaming media between users." (Final Act. 7.) In other words, the combination of O'Brien, Kobayashi, and Sylvain is the improvement of a similar device in the same way as in the prior art.

Thus, we agree with the Examiner that the combination of O'Brien, Kobayashi, and Sylvain would have rendered obvious dependent claim 15, which includes the limitation "wherein the server is configured to select the relay based on a SIP Invite message."

Dependent claims 27 and 39 recite limitations similar to those discussed with respect to dependent claim 15, and Appellants have not presented any additional substantive arguments with respect to these claims. We sustain the rejection of claims 27 and 39 for the same reasons discussed with respect to claim 15.

§ 103 Rejection—O'Brien, Jr., Kobayashi, and Mangal

We are persuaded by Appellants' arguments (Reply Br. 6) that the combination of O'Brien, Kobayashi, and Mangal would not have rendered obvious dependent claim 11, which includes the limitation "wherein the server is configured to select the relay to decrease the latency of the communication."

Claim 8 recites "a server configured to receive, process and transmit SIP signaling messages *to setup the communication* between the caller and the call destination" (emphasis added). Dependent claim 11 further recites "wherein the server is configured to select the relay to decrease the latency of *the communication*" (emphasis added). Accordingly, the limitation "setup the communication" is distinguishable over the limitation "the communication."

The Examiner found that the reduction in latency setup for the wireless communication system of Mangal corresponds to the limitation "wherein the server is configured to select the relay to decrease the latency of the communication." (Ans. 13; *see also* Final Act. 8.) We do not agree.

Mangal relates to initiating real-time media sessions, in particular, "to reduce latency in the initiation of 'instant connect' or 'push to talk' (PTT) sessions between data-capable mobile stations." (Col. 1, ll. 11–14.) Mangal explains that in one embodiment, "one way to reduce call setup latency is to buffer an initial media transmission until a link exists to transmit the media further" and "that call setup latency will normally be unnoticeable to the terminating end, as long as the terminating end ultimately receives the initial media transmission." (Col. 4, ll. 45–51.)

Although the Examiner cited to the buffering of the initial media transmission of Mangal to reduced call setup latency (Ans. 13), the Examiner has not provided sufficient evidence to support a finding that Mangal teaches the limitation "wherein the server is configured to select the relay to decrease the latency of the communication." In particular, Mangal explains that "one way to reduce call setup latency is to buffer an initial media transmission" (col. 4, ll. 45–47), rather than decreasing "the latency of the communication," as claimed. On this record, the Examiner has not demonstrated that Mangal teaches the limitation "wherein the server is configured to select the relay to decrease the latency of the communication."

Accordingly, we are persuaded by Appellants' argument that "[c]laim 11 requires the decrease of latency of the communication, not latency of the set up of the communication" and "Mangal is silent with respect to the communication." (Reply Br. 6.)

Therefore, we do not sustain the rejection of dependent claim 11 under 35 U.S.C. § 103(a).

Dependent claims 23 and 35 recite limitations similar to those discussed with respect to dependent claim 11. We do not sustain the rejection of claims 23 and 35 for the same reasons discussed with respect to claim 11.

NEW GROUND OF REJECTION UNDER 37 C.F.R. § 41.50(b)

We enter the following new ground of rejection:

Dependent claim 11 is rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien and Kobayashi.

Claim 11 recites "wherein the server is configured to select the relay to decrease the *latency* of the communication" (emphasis added). One relevant plain meaning of "latency" is "[t]he time required for a signal to travel from one point on a network to another." MICROSOFT® COMPUTER DICTIONARY 306 (5th ed. 2002). This definition of "latency" is consistent with Appellants' Specification, which states that "[t]he pre-proxy server 75 can allocate the closest RTP [Real Time Transport Protocol] relay between the two calling parties" and "[t]hat allocation enables the ability to decrease latency and travel time of the RTP stream." (Spec. 6:13–15.) Accordingly, under the broadest reasonable interpretation consistent with Appellants' Specification, we interpret "latency" as the time required for a signal to travel from one point on a network to another.

As discussed previously, O'Brien explains that "[t]he quality of a VoIP call is insured by controlling the path of the media stream to ensure that these voice packets traverse a known . . . IP network" such that "[t]he path of the voice packets can avoid congested networks." (Col. 2, ll. 26–29.) Also discussed previously, Kobayashi provides an example in which relay control section 110 uses "relay servers M**301** and M**302** . . . to check whether it is possible or not to set a path from the Web server S**1** to the QoS path reference relay control cache server C**301** without passing through the congestion section [link L1]." (¶ 131.)

Thus, both O'Brien and Kobayashi explain that congested paths (i.e., paths with heavy network traffic) should be avoided, which would result in a decrease in latency for a signal to traveling from one point on the network to another. Accordingly, the combination of O'Brien and Kobayashi teaches the limitation "wherein the server is configured to select the relay to decrease the latency of the communication," as recited in dependent claim 11.

Dependent claims 23 and 35 recite limitations similar to those discussed with respect to dependent claim 11.

Pursuant to our authority under 37 C.F.R. § 41.50(b), we reject dependent claims 11, 23, and 35 as unpatentable over O'Brien and Kobayashi under 35 U.S.C. § 103(a).

This decision contains a new ground of rejection pursuant to 37 C.F.R. § 41.50(b).

37 C.F.R. § 41.50(b) provides that a "new ground of rejection pursuant to this paragraph shall not be considered final for judicial review."

37 C.F.R. § 41.50(b) also provides that Appellants, WITHIN TWO MONTHS FROM THE DATE OF THE DECISION, must exercise one of the following two options with respect to the new ground of rejection to avoid termination of proceedings (37 C.F.R. § 1.197 (b)) as to the rejected claims:

> (1) *Reopen prosecution*. Submit an appropriate amendment of the claims so rejected or new evidence relating to the claims so rejected, or both, and have the matter reconsidered by the examiner, in which event the prosecution will be remanded to the examiner. . . .
>
> (2) *Request rehearing*. Request that the proceeding be reheard under § 41.52 by the Board upon the same Record. . . .

DECISION

The Examiner's decision rejecting claims 8–10, 12–22, 24–34, and 36–46 is affirmed.

The Examiner's decision rejecting claims 11, 23, and 35 is reversed.

A new ground of rejection has been entered under 37 C.F.R. § 41.50(b) for claims 11, 23, and 35, rejected under 35 U.S.C. § 103(a) as unpatentable over O'Brien and Kobayashi.

No time period for taking any subsequent action in connection with this appeal may be extended under 37 C.F.R. § 1.136(a)(1).

AFFIRMED-IN-PART
37 C.F.R. § 41.50(b)