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(71) Demandeurs/Applicants:
STUMM, MICHAEL, CA;
SNELGROVE, MARTIN, CA

(72) Inventeurs/Inventors:
STUMM, MICHAEL, CA;
SNELGROVE, MARTIN, CA

(74) Agent: CURRIER, T. ANDREW

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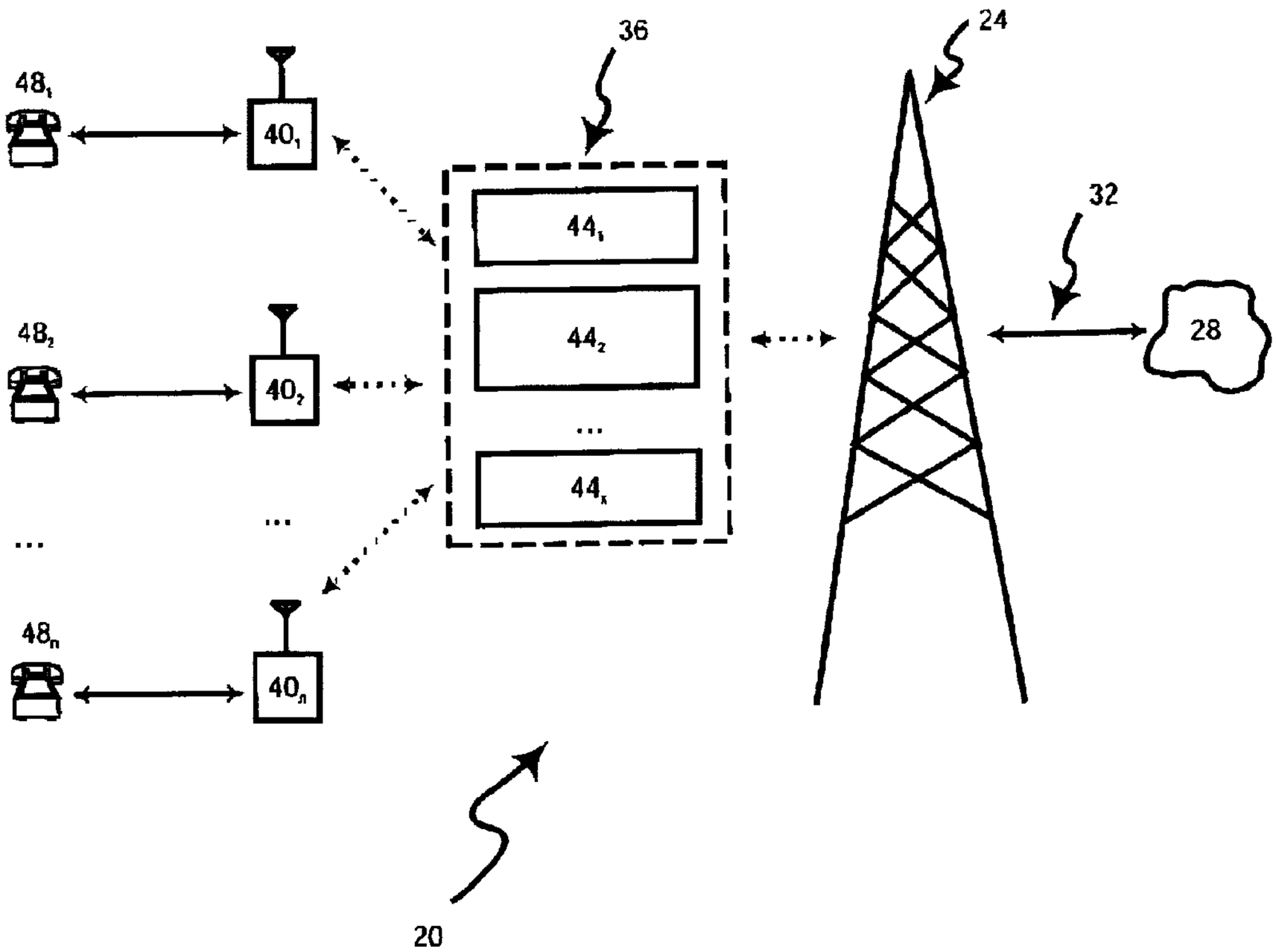


(57) Abrégé/Abstract:
The present invention provides a novel system, method and apparatus for managing call progress in a multiple access system having a base station and a plurality of subscriber stations. The system provides for a call manager at a subscriber station that is operable to generate call progress tones or other indicia to a telephone at the subscriber station, and to reserve the allocation of a voice channel between the subscriber station and the base station until the telephone at the subscriber station and a telephone at the other end of the telephone call are actually ready to connect, such as where both telephones are actually picked-up off their receiver.

ABSTRACT

The present invention provides a novel system, method and apparatus for managing call progress in a multiple access system having a base station and a plurality of subscriber stations. The system provides for a call manager at a subscriber station that is operable to generate call progress tones or other indicia to a telephone at the subscriber station, and to reserve the allocation of a voice channel between the subscriber station and the base station until the telephone at the subscriber station and a telephone at the other end of the telephone call are actually ready to connect, such as where both telephones are actually picked-up off their receiver.

CALL PROGRESS MANAGEMENT

FIELD OF THE INVENTION

The present invention relates to a system, apparatus and method of providing user-services over wireless telecommunications services, or the like. More specifically, the present invention relates to call management in a manner which can provide good usage of available radio resources in wireless systems.

BACKGROUND OF THE INVENTION

Wireless phone systems are well known and are becoming increasingly commonplace. CDMA, FDMA and TDMA are common techniques used for mobile wireless phone systems and traditionally, such wireless technology has focused on tackling problems with establishing and maintaining telephone connections where subscriber stations are mobile in relation to one or more base stations.

However, interest has also been expressed in wireless local loop (WLL) telephone systems, whereby the subscriber station remains fixed in the subscriber's premises. Known WLL systems generally implement fixed versions of known mobile wireless systems, including analog cellular and GSM or CDMA cellular. The demand for WLL systems is expected to increase in North America where local telephone service is now substantially deregulated, but the cost of wiring homes to compete with existing local telephone companies is generally prohibitive. On the other hand, WLL offers the ability to provide local telephone service without the need to wire the subscriber's house. One problem with implementing fixed versions of known mobile wireless systems is the incompatibility with subscriber's existing handsets, facsimile machines, answering machines and other equipment with such known mobile systems, thus requiring subscribers to buy new equipment. It is believed that such incompatibility will not only be a financial deterrent to the adoption of WLL systems based on known mobile systems, but will also create a marketing barrier, as subscribers wary of new technology may desire that any competing telephone service be substantially identical to their existing wired telephone system. Another factor in the implementation of wireless systems is the efficient use of radio resources, to allow, for example, a maximum number of subscribers to share a common range of bandwidth.

Furthermore, current wireless systems are not efficient in certain aspects of the management of radio bandwidth.

Overall, the present inventors have determined that, while WLL systems based upon mobile cellular technologies can provide reasonable voice performance, they do not provide a cost effective range of services and/or bandwidth efficiencies that will be desired for WLL systems.

SUMMARY OF THE INVENTION

It is an object of the present invention to provide a novel system, apparatus and method of call management via a communication link which obviates or mitigates at least one of the above-identified disadvantages of the prior art.

In a first embodiment of the invention there is provided a system for managing a communication link, comprising: a first node for linking a bandwidth-sparse medium and a second medium; a second node connectable to the first node via the second medium; and a third node connectable to the second node via the first node using a channel over the bandwidth-sparse medium, the third node having a call-manager operable to exchange signaling information with the first node via the bandwidth-sparse medium and for establishing the channel when the first node signals the call manager that the second node desires a connection to the third node.

In a particular aspect of the first embodiment, the channel is a voice channel and the signaling information occurs over a signaling channel.

In another particular aspect of the first embodiment, the second medium is a bandwidth-abundant medium, such as a public switched telephone network.

In another particular aspect of the first embodiment, the second node is a telephone handset.

In another particular aspect of the first embodiment, the first node is a wireless base station and the third-node is a wireless subscriber station connected to a telephone handset.

In another embodiment of the invention, there is provided a method for placing a call over a communication link comprising the steps of:

initiating the call using a telephone connected to a wireless subscriber station;

generating a dial-tone at the subscriber station and presenting the dial-tone at the

telephone;

receiving a telephone number at the subscriber station;

transmitting the number over a signaling channel to a base station;

initiating a wired telephone connection between the base station and a destination telephone associated with the number;

outputting a connection status indicia at the telephone corresponding to the progress of the call; and

establishing a voice communication channel between the telephone and base station for carrying a voice telephone call between the telephone and the destination telephone if each of the telephone and the destination telephone are ready to commence a voice conversation.

The present invention provides a novel system, method and apparatus for managing call progress in a multiple access system having a base station and a plurality of subscriber stations. The system provides for a call manager at a subscriber station that is operable to generate call progress tones to a telephone at the subscriber station, and to reserve the allocation of a voice channel between the subscriber station and the base station until the telephone at the subscriber station and a telephone at the other end of the telephone call are actually 'picked-up'.

BRIEF DESCRIPTION OF THE DRAWINGS

Preferred embodiments of the present invention will now be described, by way of example only, with reference to the attached Figures, wherein:

Figure 1 shows a schematic representation of a WLL system in accordance with the present invention;

Figure 2 shows a schematic representation of a subscriber station and the base station in the system of Figure 1;

Figure 3 shows a method for initiating a telephone call in accordance with another embodiment of the invention; and,

Figure 4 shows a method for receiving a telephone call in accordance with another embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

Referring now to Figure 1, a multiple access system, such as those employing TDMA, FDMA, CDMA or any other multiplexing technique, in accordance with an embodiment of the present invention is indicated generally at 20. In a present embodiment, system 20 is a wireless local loop (WLL) that includes a radio base station 24 which is preferably connected to a telecommunications network 28, such as a land line-based switched telephone network and/or a packet switched data network, by an appropriate gateway and one or more backhauls 32. Backhaul 32 can be a T1, T3, E1, E3, OC3 or other suitable land line link, or can be a satellite or other radio or microwave channel link or any other link suitable for operation as a backhaul as will occur to those of skill in the art. While backhaul 32 and telecommunications network 28 are typically used for wired connections, it is to be understood that, in general, backhaul 32 and telecommunications network 28 are conceptually a network resource that is not constrained by limited bandwidth, i.e. a bandwidth-abundant medium.

Base station 24 communicates over a communication link 36 with a plurality of subscriber stations 40 which are installed at subscriber premises. In general, it is to be understood that link 36 is conceptually a network resource that is constrained by limited bandwidth., i.e. a bandwidth-sparse medium. In a present embodiment, link 36 represents a portion of radio bandwidth dedicated to system 20. Link 36 is divided into a plurality of channels 44, that can be created using known TDMA, FDMA or CDMA techniques. In a present embodiment, each channel 44 can be dedicated to a particular subscriber station 40, as needed. As will be discussed in greater detail below, each channel 44 can occupy different amounts of bandwidth in relation to each other, in accordance with the amount of radio resources required for a particular type of connection between a subscriber station 40 and base station 24. As shown in Figure 1, there are 'x' channels, the number 'x' depending on the amount of bandwidth dedicated to link 36 and the size of each channel 44.

The number 'n' of subscriber stations 40 can vary depending upon the amount of radio bandwidth available over link 36 and/or the requirements of the subscriber stations 40. In the present embodiment, link 36 has a bandwidth capacity that is generally capable carrying fewer channels 44 than would be needed if each subscriber station 40 simultaneously required, for

example, a voice connection with base station 24. Such capacity can be determined based on known statistical models and methods for determining the likely maximum number of subscriber stations 40 that will simultaneously require a channel 44 and thereby completely occupy communication link 36.

In the illustrated embodiment of the present invention, each subscriber station 40 has at least one telephony port, such as a standard RJ-11 jack, for a conventional telephone 48. It is to be understood that additional telephony ports for facsimile and/or data can be provided in each subscriber station 40 if desired.

Figure 2 shows base station 24 and one subscriber station 40 in greater detail. Base station 24 comprises an antenna 100 for receiving and transmitting radio-communications over link 36. In turn, antenna 100 is connected to a radio 104 and a modem 108. Modem 108 is connected to a microprocessor-router assembly 112. A suitable microprocessor would be a SPARC processor system manufactured by SUN Microsystems. It will be understood that assembly 112 can include multiple microprocessors, as desired. The router within microprocessor-router assembly 112 is connected to backhaul 32 in any suitable manner, which in turn connects base station 24 to network 28, which in turn connects to at least one network subscriber telephone 120.

Subscriber station 40 comprises an antenna 200 for receiving and transmitting radio-communications over link 36. In turn, antenna 200 is connected to a radio 204 and a modem 208, which in turn is connected to a microprocessor-assembly 212.

Microprocessor-assembly 212 which can include, for example, a StrongARM processor manufactured by Intel, performs a variety of functions, including implementing A/D-D/A conversion, voice codecs, filters, encoders, data compressors and/or decompressors, packet assembly/disassembly. As seen on Figure 2, microprocessor-assembly 212 interconnects modem 208 and the previously-described telephony port 214. Accordingly, microprocessor-assembly 212 is operable to processes voice-telephone calls between telephone 48 (connected to port 214) and modem 208.

Subscriber station 40 also includes a call progress manager 216 that is connected to microprocessor assembly 212 and is operable, in conjunction with microprocessor-assembly 212 and base station 24, to manage the progress of a telephone call at subscriber station 40. Call

progress manager 216 includes software, for generating call progress indicia at the receiver of telephone 48. The generated indicia are typically audio and can be any of the known call progress tones, such as dial tone, ringing tone, busy signal, fast busy signal, stutter dial-tone. (In other embodiments of the invention, where, for example, telephone 48 is modified for use by hearing-impaired or deaf subscribers, then the output device can generate visual indicia that indicate the progress of the telephone call for the hearing-impaired subscriber. Other types of output devices can be used that are appropriate to the telephone 48 or other subscriber input-output device.)

Call progress manager 216 is also operable to manage the various states of a telephone call of (or other use of) telephone 48, and to manage the allocation/deallocation of channels 44 between subscriber station 40 and base station 24. Furthermore, call progress manager 216 is also operable to establish an appropriately sized channel 44, in accordance with the state of a telephone call at telephone 48. Further details about subscriber station 40 and, in particular, call progress manager 216 will be apparent from the discussion below.

As shown in Figure 2, antenna 100 of base station 24 and antenna 200 of subscriber station 200 are interconnected via link 36. In a present embodiment, subscriber station 40 has access to two channels 44_1, 44_2 within link 36, and can use those channels 44_1, 44_2 to carry communications between subscriber station 40 and base station 24. In particular, channel 44_1 is a signaling channel that occupies a relatively small portion of bandwidth in link 36 and is used to carry signal and/or control information between subscriber station 40 and base station 24. It is presently preferred that channel 44_1 is structured so as to occupy the minimum amount of bandwidth that necessary to maintain the state of subscriber station 40 within system 20. For example, a channel structure similar to the IS-95 access channel can be used for channel 44_1. In contrast, channel 44_2 is a voice communication channel that occupies a relatively larger portion of bandwidth than channel 44_1, the amount of bandwidth being sufficient to carry a voice telephone call within desired Quality-of-service (QoS) parameters.

In general, when telephone 48 is not in use for a voice call, then voice communication channel 44_2 remains unallocated and the bandwidth otherwise occupied thereby is available for use by other subscriber stations 40 within system 20. At the same time (i.e. when telephone 48 is not in use for a voice call) then signaling channel 44_1 is available to allow subscriber station 40 to

carry control information to base station 24, such as requesting the allocation of voice communication channel 44_2 when a voice call is to be carried between telephone 48 and base station 24. In contrast, when telephone 48 is actually being used for a voice telephone call, then signaling channel 44_1 can be unallocated while voice channel 44_2 is allocated. In this situation, any signaling between subscriber station 40 and base station 24 can be performed using the bandwidth allocated to voice channel 44_2

Figure 3 shows a method of managing an outgoing telephone call in accordance with another embodiment of the invention. For purposes of assisting in the following explanation of the method of Figure 3, reference will be made to the foregoing description of system 20 and Figures 1-2. As used herein, the term "outgoing telephone call" means a telephone call originating from telephone 48 (at subscriber station 40) to, for example, network subscriber telephone 120.

Beginning at step 300 in Figure 3, an outgoing telephone call is initiated. At this point in the method, signaling channel 44_1 is allocated for subscriber station 40 and is thus carrying signal and/or control information between subscriber station 40 and base station 24. At the same time, voice channel 44_2 is unallocated for subscriber station 40 and is thus available for use by other subscriber stations 40 within system 20. In a present embodiment, the call initiation at step 300 is completed by a subscriber using telephone 48, by "picking-up" the handset. Typically the handset is then placed to the ear of the subscriber. The removal of the handset also causes a signal to be directed to microprocessor-assembly 212 and call manager 216, which indicates that the subscriber is initiating an outgoing telephone call.

At step 310, a ready-signal is outputted. In a present embodiment, the ready-signal is a conventional dial-tone that is generated by call manager 216, and played at the earpiece of telephone 48 for the listening pleasure of the subscriber. In a present embodiment, the dial-tone can be outputted until the subscriber begins dialing digits in telephone 48 or hangs-up telephone 48 without dialing any digits. However, it is to be understood that, in other embodiments, call manager 216 can play an off-hook signal (such as a multi-frequency howl) if the handset has been off of the cradle of telephone 48 for a predetermined period of time, without the subscriber having dialed any digits or hanging up the telephone 48.

At step 320, the destination telephone number is received. This can be accomplished by

providing any form of subscriber input that indicates to subscriber station 40 a telephone number that the subscriber wishes to contact. However, in a presently preferred embodiment, this is accomplished by collecting digits that are dialed by the subscriber. Once the first digit is dialed, the dial-tone generated at step 310 is discontinued, and the subscriber only hears DTMF tones (usually generated by telephone 48) that correspond to each digit on telephone 48 as they are depressed. Call manager 216 receives the dialed digits and, using a look-up table, ensures that the dialed sequence conforms with a calling -plan of accepted telephone number sequences. (Such a look-up table can be periodically updated, as needed, by base station 24 by sending signaling messages via channel 44_1.) For example, in the North American standard, call manager 216 will look for long distance numbers in the format 1-XXX-XXX-XXXX. As is also common in the North American standard, call manager 216 can also look for numbers such as 411, 611 or 911 and connect the subscriber to the appropriate service. Such look-up tables are well known, and can be programmed into call manager 216. If the sequence of digits collected at step 320 do not conform with a recognizable sequence, then call manager 216 can output an error message to the earpiece of telephone 48, such as an audio message that states "The number you have dialed cannot be completed as dialed. Please check the number and try your call again."

However, if at step 320, a recognizable sequence of digits are received, then the sequence of digits are transmitted to base station 24 over signaling channel 44_1.

Next, at step 330 a wired connection is initiated. In a present embodiment, this occurs as base station 24 receives the sequence of digits collected at step 320, and microprocessor-router 112 uses the sequence of digits to initiate a wired connection from base station 24 and over backhaul 32, network 28 and to telephone 120. At this point, microprocessor-router 112 determines, using conventional methods, whether telephone 120 is 'busy' or 'ringing' and sends this status information back to subscriber station 40 via signaling channel 44_1. It is to be understood that the actual busy-signal tone or ringing-signal tone is not sent over link 36, but the status information is sent (typically as one or more signaling messages) to call manager 216.

Next, at step 340, a connection-status signal is outputted. In a present embodiment, the connection-status signal is a tone generated by call manager 216 and outputted at the earpiece of telephone 48. The generated tone is appropriate to the status of the information received by subscriber station 40, and thus, if the wired connection between base station 24 and telephone

120 is busy, then a busy signal is outputted to the earpiece of telephone 120. However, if the wired connection is 'ringing', then a ringing tone is outputted at the earpiece of telephone 120.

The method advances to step 350 where it is determined whether the wired connection is established. If a busy-signal was detected by base station 24, then it is determined the wired connection was not established and the method advances to step 360, the call termination step, where the wired connection is disconnected and the busy-signal is outputted from the earpiece of telephone 48 until the subscriber hangs-up telephone 48. Similarly, if the subscriber hangs-up the handset of telephone 48 prior to the answering of telephone 120, then the method advances to step 360 and the wired connection is disconnected thus terminating the entire call.

If, however, telephone 120 is 'answered' (i.e. the handset is picked up, or the call is forwarded to a voicemail service or other telephone that is picked-up), then the method advances to step 370, where the wireless voice connection is established. In a present embodiment, this is accomplished by allocating voice channel 44_2 which is connected with the wired connection established between base station 24 and telephone 120, and thus completing an entire voice connection between telephone 48 and telephone 120. If desired, signaling channel 44_1 can be unallocated during the voice call between telephone 48 and telephone 120 and freeing up this bandwidth for use by other subscriber stations 40 within system 20.

The method eventually advances from step 370 to step 360 when the call is terminated in a usual manner, at the request of either subscriber, by, for example, hanging up either telephone 48 or telephone 120. Voice channel 44_2 is unallocated and signaling channel 44_1 is reallocated.

It is to be understood that, in general, voice channel 44_2 is preferably allocated as late as possible during the method, but within acceptable QoS parameters, in order to keep as much bandwidth as possible available for use by other subscriber stations 40 within system 20 until it is known that both the originating and receiving subscribers wish to complete the call. It will also be apparent that it may be necessary to allocate voice channel 44_2 earlier in the method in order to reduce latency between the answering of the call at telephone 120 and the connection of the voice connection over channel 44_2 with telephone 48. Such latency, if any, can be set according to the desired QoS for subscriber station 40. In some situations, it can be desired to offer a different QoS to each subscriber station 40, and charge a fee-for-service accordingly. For example, a 'silver' level QoS may be willing to tolerate some latency between the answering of telephone

120 and the actual connection over channel 44_2. In contrast, a 'gold' QoS may require no latency, by allocating channel 44_2 at some point earlier in the method shown on Figure 3.

Other economic incentives for latency tolerance will occur to those of skill in the art and are within the scope of the invention. For example, where link 36 has no bandwidth available for a given subscriber, then the subscriber can be deprived of a dial-tone or provided with some other 'not-ready' signal, to indicate to the subscriber that bandwidth is currently unavailable.

It will now be understood by those of skill in the art that, where the uplink (i.e. from subscriber station 40 to base station 24) is CDMA based, then the allocation of channel 44_2 at the last possible moment has an added benefit of reducing interference between different subscriber stations 40 in system 20, thus making it easier for base station 24 to process incoming traffic on the uplink. Thus, the allocation of channel 44_2 can actually occur in two stages: First, the channel 44_2 can be reserved for use by subscriber station 40, but not actually activated to carry information between subscriber station 40 and base station 24 until the wired connection is established between base station 24 and telephone 120.

It will be further understood that, where system 20 is intended to carry both voice (or any application with less tolerance for latency) and data (or any application with more tolerance for latency), then channel 44_2 can be used to carry portions of data to other subscriber stations 40 within system 20 while the wired connection between base station 24 and telephone 120 is being established, but reserved for a voice connection using telephone 48 once the wired connection is established. Once the wired connection with telephone 120 is established, channel 44_2 can be dedicated for a voice conversation between subscriber station 40 and base station 24; the remainder of the data can be sent to other subscriber stations 40 can be sent over other channels on link 36. It is believed that foregoing can be used to reduce the time to establish the wireless connection with telephone 48 once telephone 120 is answered. In general, in such mixed data and voice networks, the tolerance for latency on data communications can be used as a buffer for switching bandwidth between voice and data usage, to thus increase efficient use of the bandwidth on link 36.

Figure 4 shows a method of managing an incoming telephone call in accordance with another embodiment of the invention. As used herein, the term "incoming telephone call" means a telephone call originating from, for example, telephone 120 to subscriber telephone 48. For

purposes of assisting in the following explanation of the method of Figure 4, reference will be made to the foregoing description of system 20 and Figures 1-2.

Beginning at step 400 in Figure 3, an incoming telephone call is received at base station 24. To assist in the explanation of the embodiment, it will be assumed that the telephone call is coming from telephone 120. At this point in the method, signaling channel 44_1 is allocated for subscriber station 40 and is carrying control and/or signaling information between subscriber station 40 and base station 24. In contrast, voice channel 44_2 is unallocated and is available for use by other subscriber stations 40 in system 20.

Next, at step 410, subscriber station 40 is notified of the incoming call. This is accomplished by sending status information over signaling channel 44_1. In a presently preferred embodiment, the actual 'ringing signal' is not sent, but simply the status information of the incoming call is sent as one or more control bits.

At step 420, call manager 216 sends a signal to telephone 48 to make it 'ring', and thus indicate to the subscriber that there is an incoming call. The method then advances to step 430, where telephone 48 continues to ring and it is determined whether telephone 48 is answered.

If telephone 48 is answered, then the method advances to step 440, at which point the wireless connection is established. In a present embodiment, call manager 216 discontinues sending a 'ring' to telephone 48, voice channel 44_2 is allocated to carry voice traffic between subscriber station 40 and base station 24. The incoming call is thus connected, as a complete voice telephone line is established from telephone 120, through base station 24, over channel 44_2 and to telephone 48. Channel 44_2 remains allocated until telephone 48 and/or telephone 120 is hung-up and the method advances to step 450 where the call is terminated by unallocating voice channel 44_2 and reallocating signaling channel 44_2.

If however, at step 430, telephone 48 is not answered before telephone 120 is hung-up, then the method advances directly step 450 without ever establishing voice channel 44_2, and the call is terminated. Where the method advances directly from step 430 to step 450, call manager 216 will discontinue ringing telephone 48 without ever having allocated voice channel 44_2, thus having preserved this bandwidth for use by other subscriber stations 40 within system 20.

While the embodiments discussed herein are directed to specific implementations of the invention, it will be understood that combinations, sub-sets and variations of the embodiments

are within the scope of the invention. For example, the present invention can work in conjunction with a voicemail system integral to system 20 in such a manner as to obtain the bandwidth management benefits discussed herein. Further bandwidth advantages can be obtained by using the present invention in conjunction with the voicemail system discussed in applicant's copending application Voicemail for Wireless Systems, Canadian Patent Application Number 2,302,460

Furthermore, it will be understood that system 20 can have additional base stations 24, as desired, where some subscriber stations 40 are within range of two or more base stations 24, and that communications between multiple base stations 24 and subscriber stations 40 can be managed using known soft-handoff techniques. Other known wireless architectures can be employed. For example, base station 24 can be multi-sectored, each sector being defined by directional antennas, each sector comprising a different reception footprint and thus allowing reuse of available spectrum between sectors serviced from a single base station 24.

It is contemplated that the present invention can be suitable for other types of multiple access systems in addition to WLL embodiments discussed herein, such as wired networks, and can be particularly suitable for any system where the management of limited bandwidth is improved by the management of at least a portion of the call occurs at the subscriber's premises. Furthermore, it is contemplated that the present invention can be applied to mobile cellular technology, including those employing CDMA, TDMA, FDMA or other multiple access techniques or combinations thereof.

It will now be apparent that the present invention can be modified to systems incorporating three-way or n-way conference calling.

It is contemplated that the present invention can be modified to telecommunication systems for the hearing impaired, where the subscriber is given call progress indicia by, for example, visual or tactile indicators instead of audio tones. It is contemplated that the exact type of indicia can be defined to include musical numbers, radio broadcasts or advertisements. In the latter example, it is contemplated that subscribers can be offered incentives, such as reduced rates for access to system 20, where it is agreed that the subscribers will have advertisements in lieu of traditional call-progress indicia.

While the embodiments discussed herein have been directed to calls between one

subscriber station 40 in system 20 and subscribers located on a wired network, it will be understood that the present invention can be used for managing calls between subscriber stations 40 within system 20.

The present invention provides a novel, system, method and apparatus for managing calls in a multiple access system by providing a subscriber station with a call manager or other means to manage the generation of call progress tones and leaving the allocation of a portion of a communication link with limited resources until it is known that an originating or incoming call will actually be completed. In a CDMA uplink, the present invention can reduce interference on the uplink from a subscriber station to a base station, thus assisting in the base stations ability to simultaneously process uplink traffic from a plurality of other subscriber stations. The present invention also presents an interface to wireless system that appears substantially identical to a wired network, thus assisting in consumer acceptance of the wireless system and providing enhancements of bandwidth management over known wireless systems. In addition, by adding functionality to the subscriber station, the scalability of the base station can be enhanced as resources added to the base station need not handle the generation of tones or other output signals at the subscriber's telephone.

The above-described embodiments of the invention are intended to be examples of the present invention and alterations and modifications may be effected thereto, by those of skill in the art, without departing from the scope of the invention which is defined solely by the claims appended hereto.

We Claim:

1. A system for managing a communication link, comprising:
a first node for linking a bandwidth-sparse medium and a second medium;
a second node connectable to said first node via said second medium; and,
a third node connectable to said second node via said first node using a channel over said bandwidth-sparse medium, said third node having a call-manager operable to exchange signaling information with said first node via said bandwidth-sparse medium and for establishing said channel when said first node signals said call manager that said second node desires a connection to said third node.

2. The system according to claim 1 wherein said channel is a voice channel and said signaling information occurs over a signaling channel.

3. The system according to claim 1 wherein said second medium is a bandwidth-abundant medium.

4. The system according to claim 3 wherein said bandwidth-abundant medium is a public switched telephone network.

5. The system according to claim 4 wherein said second node is a telephone handset.

6. The system according to claim 1 wherein said first node is a wireless base station and said third-node is a wireless subscriber station connected to a telephone handset.

7. The system according to claim 1 wherein said bandwidth-sparse medium is a wireless communication link.

8. The system according to claim 7 wherein said channel is based on a CDMA structure.

9. The system according to claim 1 wherein said third node includes a telephone handset and said call-manager is further operable to present call-progress indicia to a user of said telephone handset prior to said establishment of said channel.

10. The system according to claim 9 wherein said indicia includes a dial-tone when said user initiates a call using said handset.

11. The system according to claim 9 wherein said indicia includes a busy-signal when said second-node is unavailable for connection with said first node.

12. The system according to claim 9 wherein said indicia includes a ringing-tone when said first-node initiates a connection with said second-node and said second-node is available for connection with said first node.

13. The system according to claim 9 wherein said indicia includes a ringing said handset when said second-node is attempting to initiate a connection with said third-node.

14. The system according to claim 1 wherein said second medium is also a bandwidth-sparse medium for connecting said first node and said second node using a second channel.

15. The system according to claim 14 wherein said second node is connectable to said first node using said second channel, said second node having a call-manager for establishing said second channel; said channels being established by their respective call managers when said second-node and said third-node indicate to said first node that each desire to complete the connection between said second-node and said third-node.

16. A system for managing a communication link comprising:

a first node for linking a bandwidth-sparse medium and a second medium;

a second node connectable to said first node via said second medium; and,

a third node connectable to said second node via said first node using a channel over said bandwidth-sparse medium, said third node having a call-manager for establishing said channel when said third node and said second node are each ready to establish a connection.

17. An apparatus for managing a communication link between a first node and a second node via a third node, comprising:

a first interface for connection to said third node using one of a signaling channel and an application channel over a bandwidth-sparse medium;

a second interface for connection to a user input-output device;

a microprocessor for converting said application to appropriate formats for each of said interfaces; and

a call manager for establishing said application channel when said first node and said second node are each ready to establish a connection through said third node, said third node signaling said second node's readiness via said signaling channel.

18. The apparatus according to claim 17 wherein said application channel is a voice channel.

19. The apparatus according to claim 17 wherein said second node and said third node are connectable over a bandwidth-abundant medium.

20. The apparatus according to claim 19 wherein said bandwidth-abundant medium is a public switched telephone network.

21. The apparatus according to claim 17 wherein said second node is a telephone handset.

22. The apparatus according to claim 17 wherein said third node is a wireless base station and said first node is a wireless subscriber station and said device is a telephone handset.

23. The apparatus according to claim 17 wherein said bandwidth-sparse medium is a wireless communication link.

24. The apparatus according to claim 17 wherein said channels are based on a CDMA structure.

25. The apparatus according to claim 17 wherein said device is a telephone handset and said call-manager is further operable to present call-progress indicia to a user of said telephone handset prior to said establishment of said channel.

26. The apparatus according to claim 17 wherein said indicia includes a dial-tone when said user initiates a call using said handset.

27. The apparatus according to claim 17 wherein said indicia includes a busy-signal when said second-node is unavailable for connection with said first node.

28. The apparatus according to claim 17 wherein said indicia includes a ringing-tone when said third node initiates a connection with said second-node and said second-node is available for connection with said first node.

29. The apparatus according to claim 17 wherein said third node and said second node are connectable over a bandwidth-sparse medium using a second channel.

30. The apparatus according to claim 29 wherein said second node is connectable to said third node using said second channel, said second node having a call-manager for

establishing said second channel; said channels being established by their respective call managers when said second-node and said first-node indicate to said first node that each are ready to connect.

31. A base station comprising:

a first interface for connecting said base station to a first node via a bandwidth sparse medium using an application channel;

a second interface for connecting said base station to a second node via a second medium;

a processing unit for passing said application between said interfaces, said processing unit operable to, in cooperation with said first node, to establish said application channel when said nodes indicate that each are ready to connect.

32. A method for placing a call over a communication link comprising the steps of:

initiating said call using a telephone connected to a wireless subscriber station;

generating a dial-tone at said subscriber station and presenting said dial-tone at said telephone;

receiving a telephone number at said subscriber station;

transmitting said number over a signaling channel to a base station;

initiating a wired telephone connection between said base station and a destination telephone associated with said number;

outputting a connection status indicia at said telephone corresponding to the progress of said call; and

establishing a voice communication channel between said telephone and base station for carrying a voice telephone call between said telephone and said destination telephone if each of said telephone and said destination telephone are ready to commence a voice conversation.

33. The method according to claim 32 wherein said destination telephone is ready to

commence said voice conversation when said destination telephone is answered.

34. The system according to claim 9 wherein said indicia are audible.

35. The system according to claim 9 wherein said indicia are tactile.

36. The system according to claim 9 wherein said indicia are visual.

37. The system according to claim 9 wherein said indicia are user-defined.

38. The system according to claim 34 wherein said user-defined indicia is a dial-tone that comprises at least a portion of a musical number.

39. The system according to claim 34 wherein said user-defined indicia is a dial-tone that comprises a radio broadcast.

40. The system according to claim 34 wherein said indicia is a comprises at least a portion of an advertisement.

41. The system according to claim 40 wherein said channel cannot be established until said portion of said advertisement has been completely presented.

42. The system according to claim 41 wherein said subscribers to system 20 are offered an incentive in consideration for their exposure to said advertisement.

43. A method for receiving a call from an originating telephone at a subscriber station via a communication link comprising the steps of:

receiving an incoming call for said subscriber station at a base station respective to

said subscriber station;

signaling said subscriber station of said incoming call;

generating an indicia at said subscriber station indicating said incoming call; and,

establishing a voice communication channel between said telephone and base station for carrying said incoming call between said telephone and said originating telephone if each of said telephone and said originating telephone are ready to commence a voice conversation.

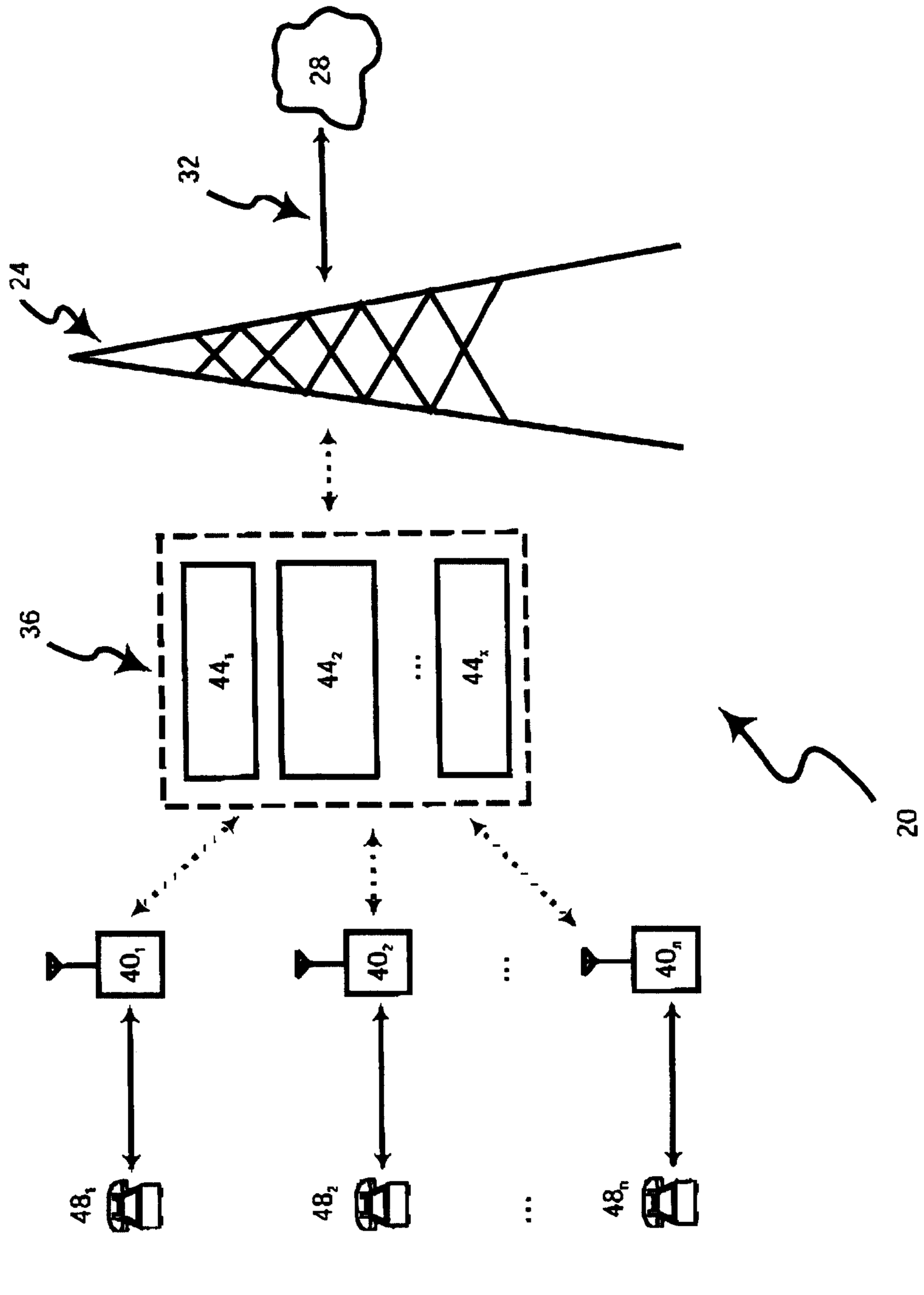


Fig. 1

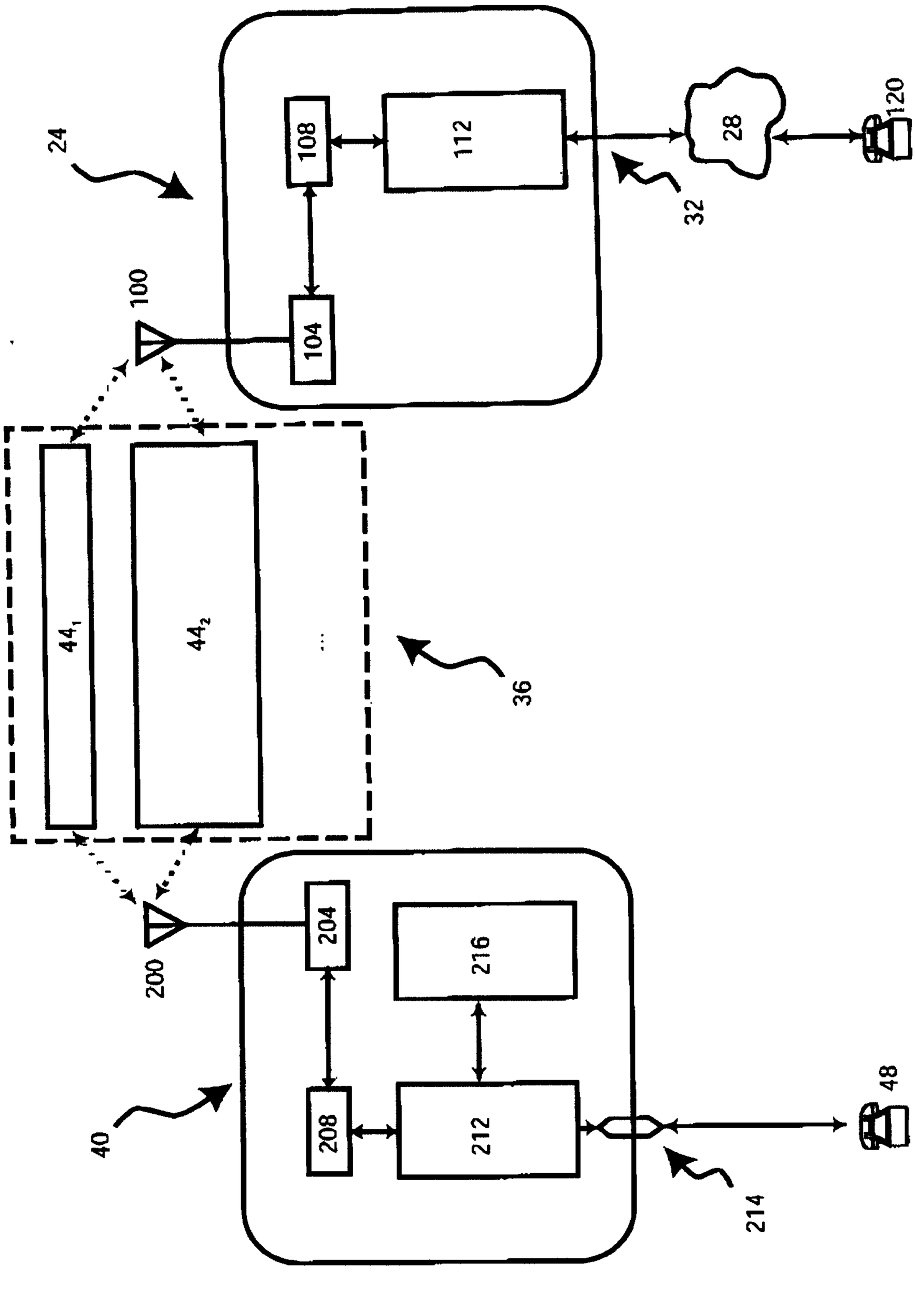

24
100
108
112
28
120
32
104
44_1
44_2
...
36
200
204
216
208
212
40
214
48

Fig. 2

Fig. 3

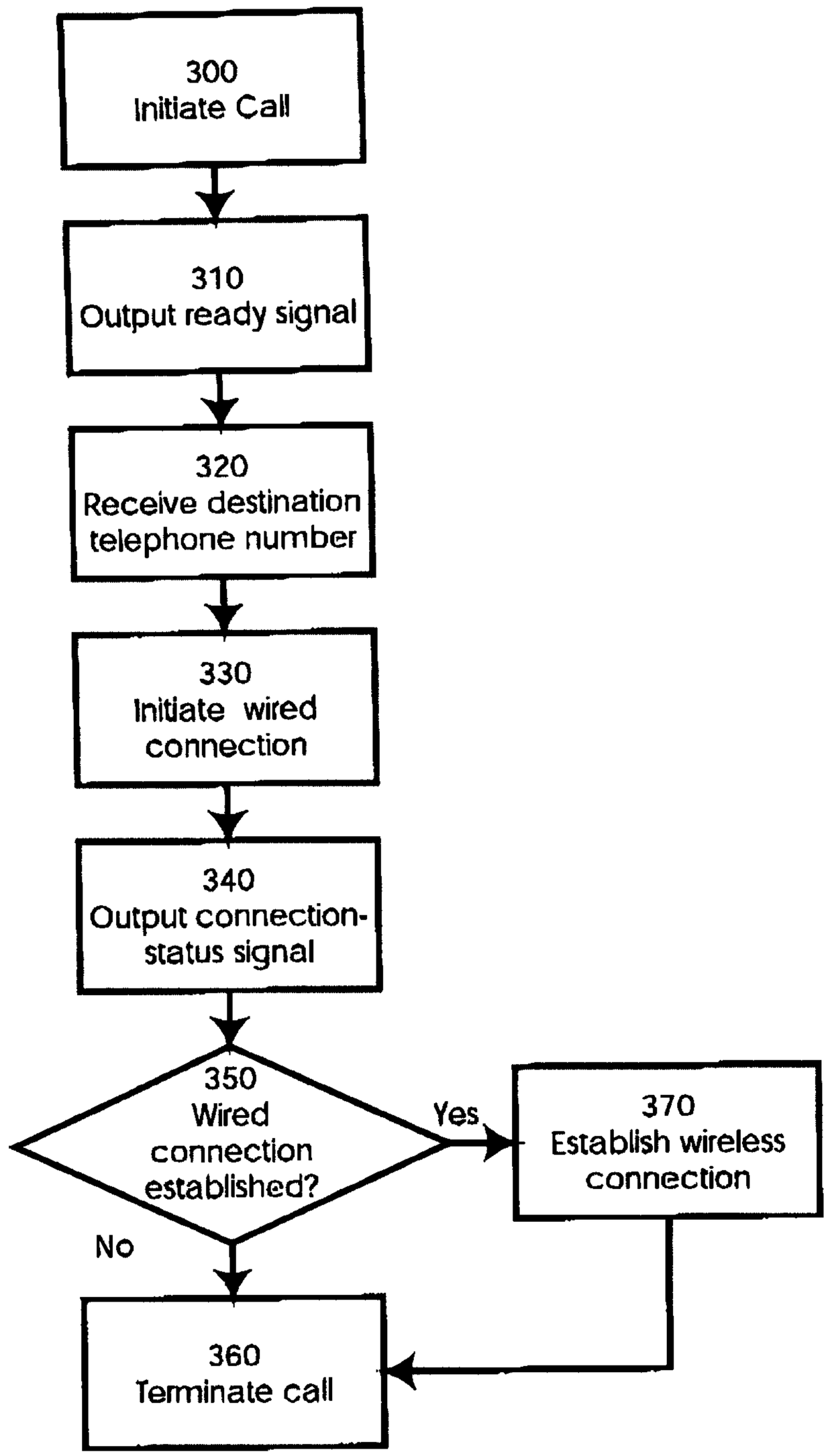
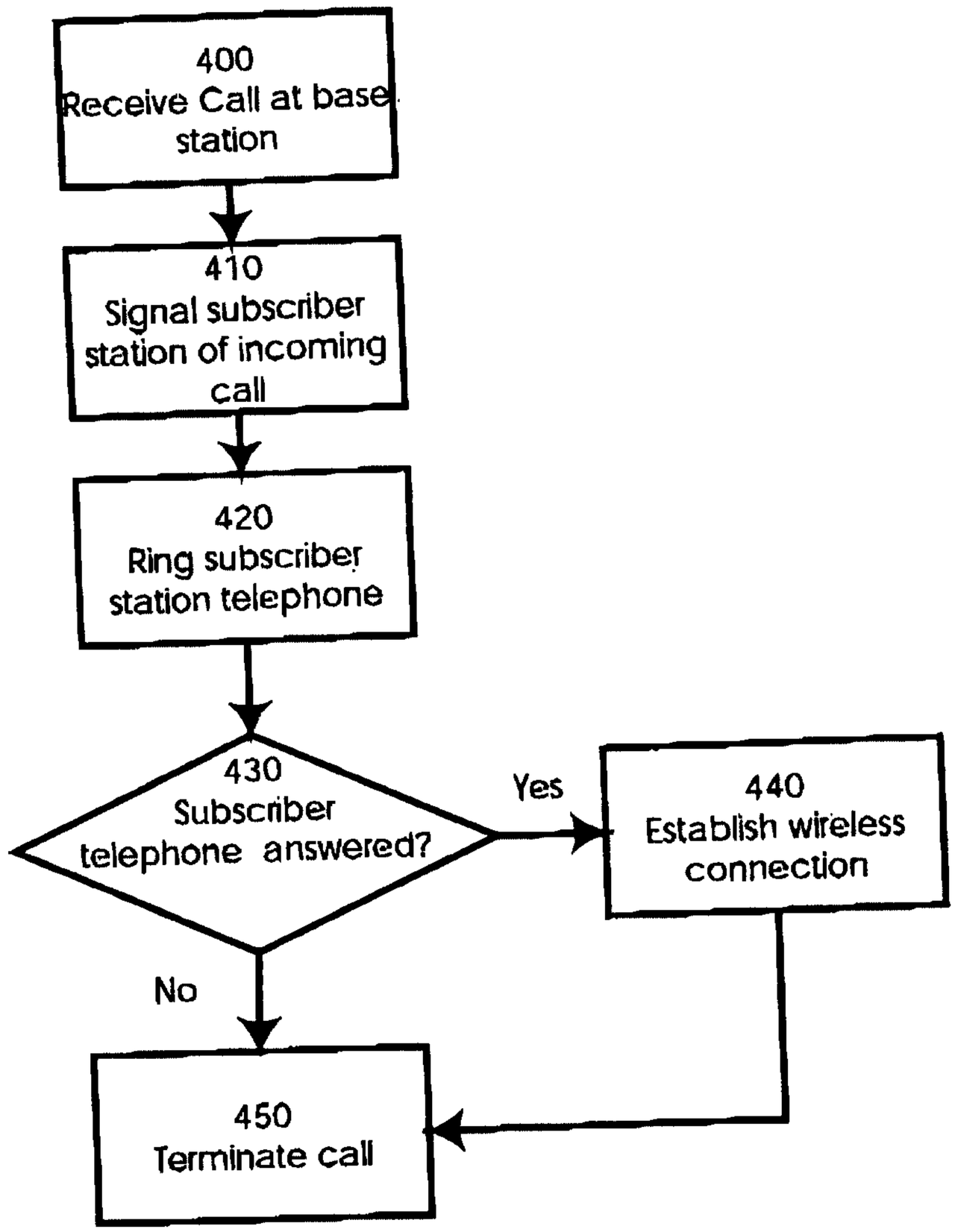
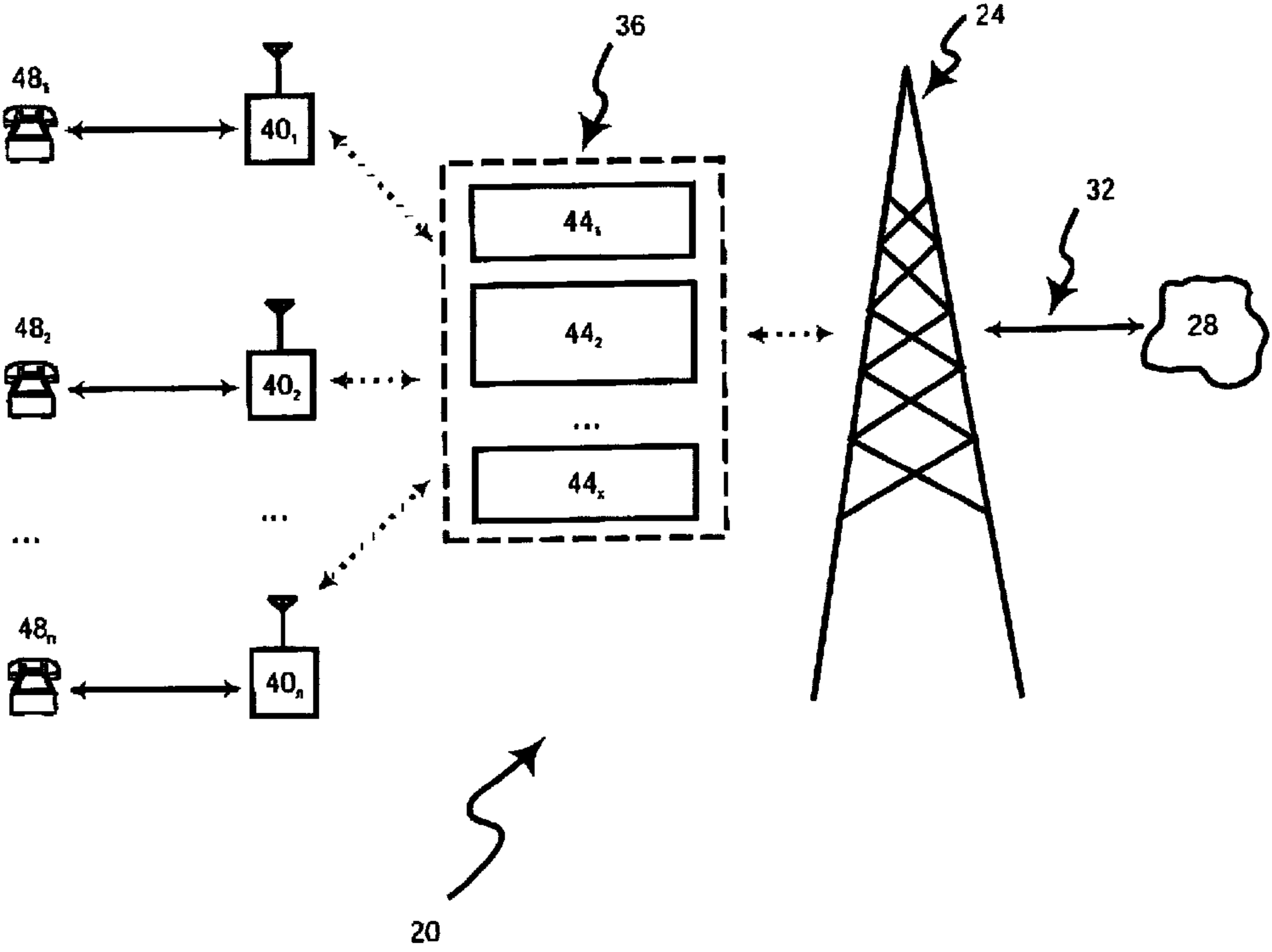

300
Initiate Call
310
Output ready signal
320
Receive destination telephone number
330
Initiate wired connection
340
Output connection-status signal
350
Wired connection established?
Yes
No
370
Establish wireless connection
360
Terminate call

Fig. 4


400
Receive Call at base station
410
Signal subscriber station of incoming call
420
Ring subscriber station telephone
430
Subscriber telephone answered?
Yes
440
Establish wireless connection
No
450
Terminate call


36
24
32
28
48_1
40_1
48_2
40_2
48_n
40_n
44_1
44_2
44_x
20