EXHIBIT C-4

Invalidity of 11,921,355 in view of Digital Hearing Aid Systems

On information and belief, systems comprising a digital hearing aid, receiver, and/or remote transmitter ("Digital Hearing Aid systems" or "DHA systems") were sold, on–sale, in public use, or otherwise publicly available prior to the earliest priority date of U.S. Patent No. 11,921,355 ("'355 patent") and qualify as prior art to the '355 patent at least under pre-AIA 35 U.S.C. §§ 102(a) and/or (b). DHA systems anticipate or, alone or with other references, render obvious, one or more of the Asserted Claims of the '355 patent.

The chart below illustrates how exemplary DHA system render the Asserted Claims of the '355 patent invalid as anticipated or obvious over the DHA systems alone or in view of additional secondary references, including but not limited to the references listed below and prior art identified in Defendant's Invalidity Contentions. One exemplary DHA system identified in this claim chart includes, for example, the Phonak Perseo digital hearing aid, MicroLink receiver, used with the Smartlink SX transmitter ("Phonak system"), which were sold, on–sale, in public use, or otherwise publicly available at least as early as March 2004. The Phonak system is evidenced by, for example, the Phonak Perseo Brochure, Phonak SmartLink SX User Guide, Phonak – SmartLink RX – FM Systems Brochure, and Phonak – MicroLink User Guide, which published at least as early as the earliest claimed priority date of the '355 patent, as well as the additional information cited herein.

The citations and descriptions provided herein refer to the Phonak system as a representative system demonstrating the relevant prior art features and functionality of the DHA systems, and is not intended to limit the invalidity contentions to this specific implementation. On the contrary, as demonstrated herein, components of the DHA systems, including the receiver and transmitter, could be used interchangeably with different hearing aids or compatible systems. Additionally, the hearing aid device could have an integrated receiver within a unified assembly. Accordingly, the chart identifies other examples of exemplary components that could be substituted for or used interchangeably with the Microlink receiver and/or Smartlink SX transmitter, including but not limited to the Oticon DigiFocus Compact digital hearing aid. Moreover, and as discussed in greater detail in the following chart, the prior art DHA systems may comprise one or more of the following components which could be used interchangeably and include substantially similar structures and functionality:

- Digital Hearing Aids: Phonak: Perseo, Perso with PersonalLogic, Claro 11, Claro 22, Claro 21dAZ, Claro dAZ, Claro dAZ FM, Supero 411, Supero 412, PicoForte3 SC-D, PicoForte3, SuperFront, Clasica, Aero 11, Aero 22, Aero 23 mZ, Aero 33, MicroZoom P2 AZ, Astro SC, Astro MC, Aero 211, Aero 211 AZ, Aero 311, Aero 311 AZ, Supero 412, Supero 413 AZ, Piconet 2 P2, Piconet P2 AZ, SonoForte P3 AZ, PowerZoom P4 AZ, NovoForte E3, NovoForte E4, Conversa, Unison; Oticon, Inc.: DigiFocus Compact, Syncro; Starkey Hearing Technologies: Genesis 4, Genesis DX with one or more of Precision Audio Imaging, Axent II, Sequel, Endeavour, Arista; Siemens Hearing Instruments, Inc.: Acuris with e2e wireless, Triano, Signia Select, Prisma 2, Music, Music Digital, Phoenix, Phoenix Pro, Infiniti 3; GN ReSound: ReSound Air; Sonic Innovations Inc.: Natura 3 with DIRECTIONALplus, Natura 2 SE, Altair, Tribute, Adesso; Widex: Senso Diva; Micro-Tech: Denali, Caballo, Dx3; Microsound A/S: Microsound Pilot with Eartunner; University Wisconsin + Nicolet Instrument Corp.: Project Phoenix; and any other prior art digital hearing aids devices, including, but not limited to devices offered by the following manufacturers: Argosy, Audifon, Beltone, Danavox, Hanaston, Interton, Oticon Philips, Phonak, Rexton, Siemens, Sonic Innovations, Unitron, and Widex.
- Receivers: Phonak: MicroLink (MLx S, ML 8, MLxi, ML3-ML7), MicroVox RX2, EduLink
- Transmitters: Phonak: SmartLink, SmartLink SX, Campus S, ClearCall, SoundPilot FM, WatchPilot Classic, WatchPilot Elegance, MAXX, CrosLink, HandyMic

- On information and belief, other configurations, models, or embodiments may also satisfy one or more claim limitations, and the inclusion of the examples identified in this chart do not preclude the existence or applicability of alternative implementations. Defendant reserves the right to supplement, amend, or otherwise update these contentions as additional information becomes available, including but not limited to information obtained through discovery.

To the extent IngenioSpec contends the DHA systems do not disclose one or more limitations of the claims, it would have been obvious to combine the DHA systems with the knowledge of a person of ordinary skill in the art ("POSITA") and/or with one or more of the following references to render the Asserted Claims the '355 patent invalid:

- U.S. Patent No. 5,721,783 ("Anderson" or "Anderson '783") was filed on June 7, 1995, and issued on February 24, 1998, and is therefore prior art under 35 U.S.C. §§ 102 (a), (b), and (e).
- U.S. Patent No. 7,245,730 ("Enzmann" or "Enzmann '730") was filed on January 13, 2003, published on July 15, 2004, issued on July 17, 2007, and is thus prior art under at least 35 U.S.C. §§102(a) and (e). Enzmann '730 issued from U.S. Application No. 10/341,293, which was published as U.S. Patent Publication No. 2004/0136555 ("Enzmann '555"). Enzmann '730 and Enzmann '555 both qualify as prior art and have corresponding disclosures, and citations herein to Enzmann '730 includes and incorporates by reference citations to the corresponding disclosures in Enzmann '555.
- U.S. Patent No. 7,150,526 ("Jannard" or "Jannard '526") was filed on July 28, 2003 and is thus prior art under at least 35 U.S.C. §§102(a) and (e).
- U.S. Patent No. 7,187,960 ("Abreu" or "Abreu '960") was filed on April 22, 2003, claims priority to a provisional application filed on April 22, 2002, and published on March 25, 2004, and is thus prior art under at least 35 U.S.C. §§102(a) and (e).
- U.S. Patent No. 7,620,433 ("Bodley" or "Bodley '433") was filed on January 15, 2003, published on July 15, 2004, and issued on November 17, 2009, and is thus prior art under at least 35 U.S.C. §§102(a) and (e).
- U.S. Patent No. 7,046,230 ("Zadesky" or "Zadesky '230") was filed on July 1, 2002, published on April 24, 2003, and issued on May 16, 2006, and is thus prior art under at least 35 U.S.C. §§102(a), (b), and (e).
- U.S. Patent No. 6,671,379 ("Nemirovski" or "Nemirovski '379") was filed on July 31, 2001, and issued on December 30, 2003, and is thus prior art under at least 35 U.S.C. §§102(a), (b), and (e).
- U.S. Patent Publication No. 2005/0058313 ("Victorian" or "Victorian '313") was filed on September 11, 2003, and published on March 17, 2005, and is thus prior art under at least 35 U.S.C. §§102(a) and (e).
- WIPO Publication No. 2004/093490A1 ("Hollemans" or "Hollemans '490") was filed on April 15, 2004, and claims priority to a PCT application filed on April 18. 2003, and is thus prior art under at least 35 U.S.C. §§102(a) and (e).
- U.S. Patent No. 6,944,474 ("Rader" or "Radar '474") was filed on September 20, 2001, published on April 3, 2003, issued on September 13, 2005, and is thus prior art under at least 35 U.S.C. §§ 102(a), (b), and (e). Rader '474 issued from U.S. Application No. 09/957,344, which was published as U.S. Patent Publication No. 2003/0064746 ("Rader '746"). Moreover, WIPO Publication No. 03/026349 ("Radar '349") is an international application publication that claims priority to the same application that issued as Rader '474. Radar '349, Rader '746, and

Rader ’474 all qualify as prior art and have corresponding disclosures, and citations herein to Radar ’474 includes and incorporates by reference citations to the corresponding disclosures in Rader ’746 and Rader ’349.

- U.S. Patent Publication No. 2005/0136839A1 (“Seshadri ’839” or “Seshadri”) was filed on October 27, 2004, claims priority to a provisional application filed on May 28, 2003, and published on June 23, 2005, and is thus prior art under at least 35 U.S.C. §102(e).
- U.S. Patent Publication No. 2005/0226446A1 (“Luo ’446” or “Luo”), filed on April 8, 2004 and published on October 13, 2005, and is thus prior art under at least 35 U.S.C. §102(e).
- U.S. Patent No. 7,010,332B1 (“Irvin ’332” or “Irvin”), filed on February 21, 2000, was patented on March 7, 2006, and is thus prior art under at least 35 U.S.C. §102(e).
- U.S. Pat. No. 6,373,942 (“Braund”) was filed on April 7, 2000 and issued on April 16, 2002, and is thus prior art under at least 35 U.S.C. §§ 102 (a), (b), and (e).
- U.S. Pat. No. 7,979,035 (“Griffin”) was filed on Dec. 4, 2007 with a priority claim to Nov. 7, 2000, and is thus prior art under at least 35 U.S.C. § 102 (e).
- U.S. Pub. No. 2003/0068057 (“Miller”) was filed on Oct. 6, 2001 and published on Apr. 10, 2003, and is thus prior art under at least 35 U.S.C. §§ 102 (a), (b), and (e).
- U.S. Pat. No. 4,955,729 (“Marx”) was filed on Mar. 29, 1988 and issued on Sep. 11, 1990, and is thus prior art under at least 35 U.S.C. §§ 102 (a), (b), and (e).
- U.S. Pat. No. 7,542,012 (“Kato”) was filed on June 16, 2005 with a priority claim to Dec. 22, 2003 and issued on Jun. 2, 2009, and is thus prior art under at least 35 U.S.C. § 102(e).

For any claim limitation, Defendant may rely on excerpts cited for any other limitation and/or additional excerpts not set forth fully herein to the extent necessary to provide further explanation for a reference’s disclosure of a limitation. Where an excerpt refers to or discusses a figure or figure items, that figure and any additional descriptions of that figure should be understood to be incorporated by reference as if set forth fully herein.

With respect to the proposed modifications to the DHA system, such modification would have been obvious to try, an obvious combination of prior art elements according to known methods to yield predictable results, a simple substitution of one known element for another to obtain predictable results, a use of known techniques to improve a similar devices or method in the same way, an application of a known technique to a known device or method ready for improvement to yield predictable results, a variation of a known work in one field of endeavor for use in either the same field or a different one based on design incentives or other market forces with variations that are predictable to a POSITA, and/or obvious in view of teachings, suggestions, and motivations in the prior art that would have led a POSITA to modify or combine the prior art references.

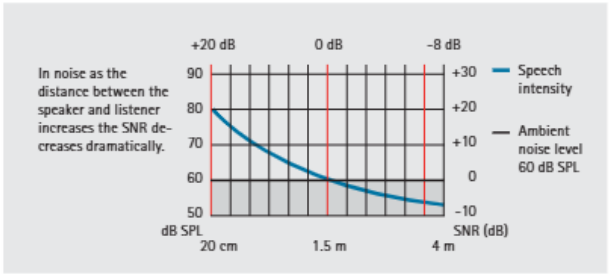
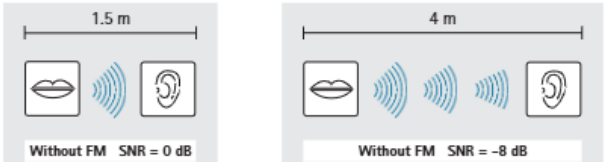
Defendant expressly reserves the right to rely on additional prior art references to demonstrate that certain features were well known in the art and would have been obvious to a POSITA as of the effective filing date of the ’355 patent, e.g., to the extent the references cited in the accompanying chart are argued or found not to expressly teach or suggest such features. For example, use of directional microphones in headworn devices, e.g., for voice recognition, receipt of audio commands, sensing of environmental sounds, and noise cancellation were well known features, and it would have been obvious to a POSITA to combine them with those of DHA systems for the reasons set forth below. For example, Braund discloses well-known aspects of using a microphone for voice recognition (see, e.g., Braund at 3:47-58, 6:23-27, 8:55-61) and rechargeable battery features (see, e.g.,

Braund at 9:6-7) as recited in at least claims 54, 64, 67, 79, and 88. Griffin discloses well-known aspects of directional microphones for noise cancellation (see, e.g., Griffin at 4:21-39,) and rechargeable battery features (see, e.g., 4:21-5:8, 9:26-33) as recited in at least claims 7, 8, 11, 12, 15, 29, 30, 33-40, 54, 57, 60, 64, 65, 67-70, 77, 79, 80, 81, 82, 87, 88-82, 100. Miller discloses well-known aspects of directional microphones for noise cancellation (see, e.g., Miller at paras. [0044]-[0045], [0062]) as recited in at least claims 7, 8, 11, 12, 15, 29, 30, 33-40, 54, 57, 60, 65, 67-70, 77, 79, 80, 81, 82, 87, 88-82, and 100. Other well-known features include use of sensors in, on, or with headworn devices, e.g., to determine if/when headworn devices were being worn. For example, Marx discloses well-known aspects of using sensor data to determine if a device is being worn and modifying operation of the device based on that determination (see, e.g., 3:34-4:12) as recited in at least claims 13, 16, 25, 26, 41, 42, 43, 44, 64, 65, 66, 78, 87. For further example, Kato discloses well-known aspects of including an accelerometer in a head worn device (see, e.g., 35:3-9) as recited in at least claim 63.

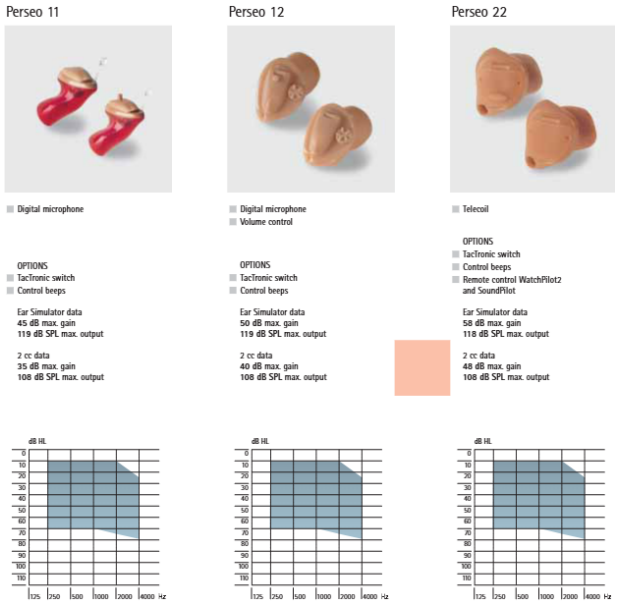
To the extent that these Invalidity Contentions rely on or otherwise embody particular interpretations of terms or phrases in the asserted claims, Defendant is not proposing any such interpretations as proper interpretations or constructions of those terms or phrases. Various positions put forth in this document are predicated on IngenioSpec's apparent interpretation of its claims as evidenced by its Infringement Contentions. Those positions are not intended to and do not necessarily reflect Defendant's interpretation of the true and proper scope of the asserted claims. Defendant further incorporates by reference all reservations of rights set forth in the Invalidity Contentions, as if fully set forth herein.

Defendant's arguments throughout these invalidity contentions are based on IngenioSpec's apparent interpretation of the claims in its infringement contentions. Defendants do not agree that such interpretation is proper and reserve the right to argue that the mapped actions do not infringe any claims of the '355 patent. In the alternative, should the Court or factfinder construe and/or apply the claims in a manner consistent with IngenioSpec's interpretations in its infringement contentions for the '355 patent, the claims are invalid for at least the reasons set forth herein.

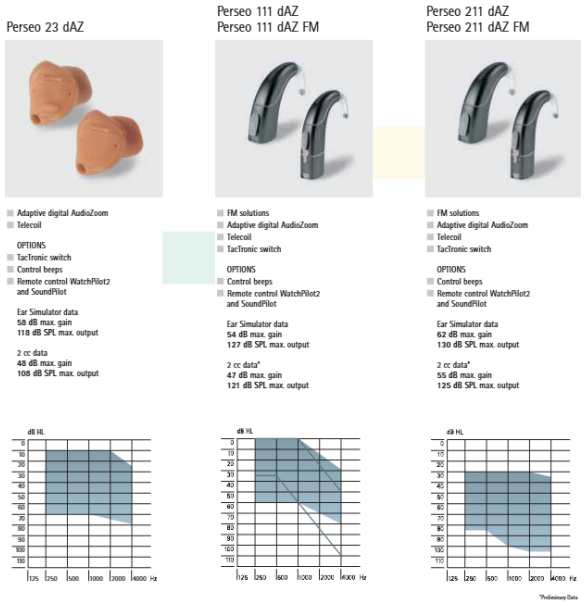
Claims	**Prior Art**
1[pre]. A head-worn personal electronic apparatus operable to at least output customized audio output, comprising:	DHA systems, alone or in view of Anderson or Enzmann or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus operable to at least output customized audio output."

Claims	Prior Art
	Phonak – Perseo Brochure at Page 5 The Science of PersonalLogic™ Phonak has applied knowledge from new research and past experience to add a new dimension of individualized logic to automatic hearing systems – PersonalLogic. Combining new and proven features, PersonalLogic optimizes the functioning, operation and fitting of hearing systems for each individual user. Thanks to PersonalLogic, Perseo takes customization to a level beyond "custom-fitted" related to individual hearing loss. The new digital system from Phonak is the first hearing system with customized automatic operation, as opposed to operation based on factory set, unchangeable rules. Now, Perseo gives hearing professionals control over automatic operation, allowing them to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes. Phonak – Perseo Brochure at Page 15 The perfectly integrated communication system **FM MicroLink** FM technology has allowed thousands of children to participate in standard education settings and to be actively involved in social activities. Phonak pioneered this development and the integration of FM into hearing instruments to produce communication systems that are effective in even the most demanding situations. Since the innovation of miniaturized Phonak **MicroLink** systems, FM has become an important solution also for adults with a high communication need in work and leisure environments. All Perseo BTEs have been designed to incorporate MicroLink with its fully integrated appearance and operation. This integrated system is unique to Phonak. **Speech-to-Noise Ratio (SNR) as a function of distance between speaker and listener**  **Speech-to-Noise Ratio without FM is distance dependent** 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 26



Claims	Prior Art
	Phonak – Perseo Brochure at Page 27



IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 7

Claims	Prior Art
	Phonak – Perseo Brochure at Page 28  Phonak – Microlink User Guide at p. 24 MicroLink MLx can be used with Behind-The-Ear (BTE) hearing instruments from several different manufacturers. Phonak – Microlink User Guide at p. 52 The HandyMic and MicroVox TX2 are used by hearing impaired people. To operate the HandyMic and MicroVox TX2, hearing instruments with FM receivers (e.g., MicroLink) are required. The user benefit of FM is an improved signal-to-noise ratio which leads to an optimized speech understanding even in very difficult listening situations.

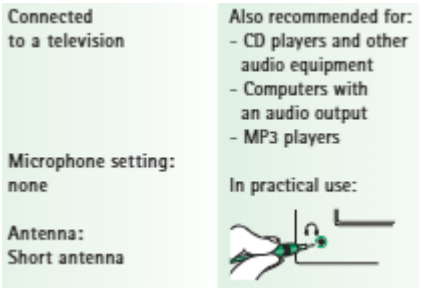
Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 In addition to the very successful launch of Phonak's new premium product line Perseo in March 2003, the following is a summary of the key product launches during the past fiscal year: • MAXX and Amio, Phonak's new entry level digital products, launched in June 2003; • Conversa, Unitron Hearing's mid-level digital product line launched in June 2003; • Supero +, an enhanced version of Phonak's digital high power line, launched in October 2003; • MAXX directional, which adds directional signal processing to the entry-level product family, launched in October 2003; and • Liaison, Unitron Hearing's new high-end product line, launched in October 2003. Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Phonak – Online Interview re Perseo (April 7, 2003) at p. .5 **AO/Beck**: Phonak has been a leader in FM technology-is this available for the Perseo product line? **Dave Fabry/Phonak**: Yes, FM can be an integrated solution for the Perseo BTEs and is compatible with all Phonak transmitters.

Claims	Prior Art
	Phonak – Online Interview re Perseo (April 7, 2003) at Page 5 In a nutshell, Perseo is designed for new and experienced users who are looking for a hearing instrument with advanced adaptive features, enhanced sound quality, easy personalization of fitting and choice of user operation (namely, automatic, manual program control with Tactronic switch, or via remote control). The choice of one product over another is dependent on the patient's hearing loss, listening needs, and prioritization of size, cost, and available performance features. Phonak provides an entire family of digital options from Perseo, Claro, Aero, and our new digital value line that will be introduced this summer. In addition, Supero is the only digital hearing instrument designed specifically for super-power applications. Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink, based on Phonak breakthrough digital FM synthesizer technology, now enables people with a hearing impairment and those with normal hearing to use the same communication tools with equal effectiveness. SmartLink is a communication gateway to all situations where understanding perfectly, all of the time, is key. The unique innovation, SmartLink, meets the needs of people with hearing loss, giving them access to effective communication in difficult situations and to the many advantages off modern communication tools, via one device.

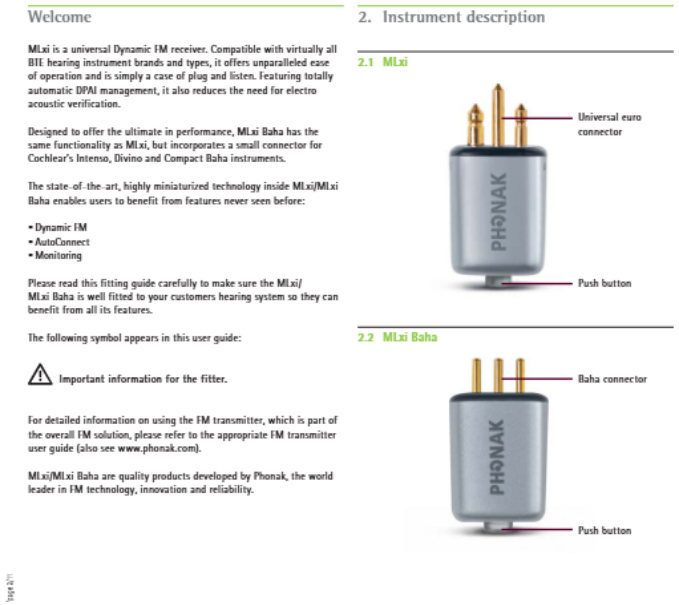
Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 4 

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 10wire  0

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 14  Phonak - SmartLink SX User Guide at p. 6 This unique combination of an FM transmitter, mobile phone link and hearing instrument remote control is the gateway to a completely new hearing experience.

Claims	Prior Art
	Phonak - SmartLink SX User Guide at p. 32  Phonak - SmartLink SX User Guide at p. 47  Phonak - SmartLink SX User Guide at p. 58 The SmartLink is used by hearing-impaired people. To operate the SmartLink, BTE hearing instruments with MicroLink receivers are required.

Claims	Prior Art
	Phonak – Fitting Guide for MLxi and MLxi Baha at Page 2



Welcome

MLxi is a universal Dynamic FM receiver. Compatible with virtually all BTE hearing instrument brands and types, it offers unparalleled ease of operation and is simply a case of plug and listen. Featuring totally automatic DPAI management, it also reduces the need for electro acoustic verification.

Designed to offer the ultimate in performance, MLxi Baha has the same functionality as MLxi, but incorporates a small connector for Cochlear's Intenso, Divino and Compact Baha instruments.

The state-of-the-art, highly miniaturized technology inside MLxi/MLxi Baha enables users to benefit from features never seen before:

- Dynamic FM
- AutoConnect
- Monitoring

Please read this fitting guide carefully to make sure the MLxi/ MLxi Baha is well fitted to your customers hearing system so they can benefit from all its features.

The following symbol appears in this user guide:

Important information for the fitter.

For detailed information on using the FM transmitter, which is part of the overall FM solution, please refer to the appropriate FM transmitter user guide (also see www.phonak.com).

MLxi/MLxi Baha are quality products developed by Phonak, the world leader in FM technology, innovation and reliability.

2. Instrument description

2.1 MLxi

2.2 MLxi Baha

Phonak – Fitting Guide for MLxi and MLxi Baha at Page 9



Dynamic FM transmitters

inspiro

inspiro is a Dynamic FM transmitter for teachers and suitable for use in both regular and special schools.

DynaMic

DynaMic is the passaround microphone for the MultiTalker Network with inspiro.

SmartLink+

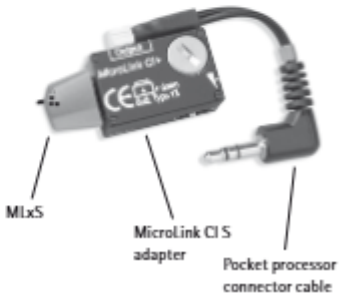
SmartLink+ is the FM transmitter for those who want nothing less than the very best.

ZoomLink+

ZoomLink+ is the FM transmitter for those who know what they want.

EasyLink+

EasyLink+ is the FM transmitter that is easiest to operate.

Claims	Prior Art
	Phonak MicroLink MLxS Product Information at Page 1 The MicroLink MLxS is the first miniaturized FM receiver with a digital frequency synthesizer for full multi-frequency flexibility. The receiver features fully automatic, non-contact frequency synchronization with either a WallPilot or the Campus S. The MicroLink MLxS is compatible with all Phonak FM transmitters and the WallPilot, and is recommended for use with this equipment. **MLxS with BTE instrument**  **MLxS with adapter for CI pocket processor**  Phonak MicroLink MLxS Product Information at Page. 1 • Compatible with most behind-the-ear hearing aids on the market

Claims	Prior Art
	Phonak SmartLink SX Product Information at p. 1 Product Information SmartLink™ SX  SmartLink SX is a compact communication device for people with hearing loss who lead busy, active lives. The SmartLink SX incorporates an FM transmitter with digital signal processing, a remote control for Phonak hearing instruments, and Bluetooth for communication with mobile phones. With its digital frequency synthesizer, the SmartLink allows a highly flexible choice of operating frequency. For maximum customer benefit, we recommend using a Phonak multi-channel FM receiver (e.g. the MLxS or ML8S) that can be automatically synchronized from the SmartLink. Phonak - Hearing Aid Museum re SmartLink SX FM System at Page 1 The SmartLink SX was a three-in-one device: an FM transmitter, a mobile phone bluetooth link and hearing aid remote control. The SmartLink SX worked with all Phonak's hearing aids beginning with the Perseo, Claro and Supero.

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 29



Phonak's FM solutions:
For maximum speech intelligibility

Whenever even latest-generation hearing instruments approach their physical limits – noise, reverberation, or distance – count on Phonak's FM solutions to deliver maximum speech intelligibility.

The frequent difficulties encountered by hearing instrument users when watching TV and making phone calls can be overcome with Phonak's FM technology.

Phonak FM solutions improve the SNR by 12 to 20 dB.

The FM systems by Phonak can be used with the majority of commercially available BTE instruments and cochlear implants.

Recommended applications:

Dialog:
For example at home, on a busy street, at the shopping mall or in a restaurant.

Group discussions:
For example at a family get-together, in a very noisy restaurant, at business meetings or conferences.

Making telephone calls:
The Phonak FM system provides access to the many capabilities of modern telecommunications devices.

Watching TV and listening to music:
Phonak FM provides access to the entire spectrum of today's information and entertainment technologies.

Schools and continuing education:
For example in classes at schools and universities, on outings and excursions, in lectures or on museum tours.

Transportation:
For example in cars, trains, trams and buses.

Sports of all kinds:
For example when cycling, playing golf, skiing, jogging or hiking.

Events:
For example at the theater, in concerts, at the movies, in church services or at community meetings.

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 30

MLx, the most flexible FM receiver ever made, can be used not only with Phonak BTEs but also with most third-party hearing instruments available on the market today. This is possible thanks to the standard Euro connector which simply plugs into the audio shoe of the hearing instrument.

Once again, Phonak demonstrates its commitment to innovative developments that serve hearing-impaired adults and children.

MLx compatible BTE makes:

- Argosy
- Audifon
- Beltone
- Danavox
- GN ReSound
- Hansaton
- Interton
- Oticon
- Philips
- Phonak
- Rexton
- Siemens
- Sonic Innovations
- Unitron
- Widex





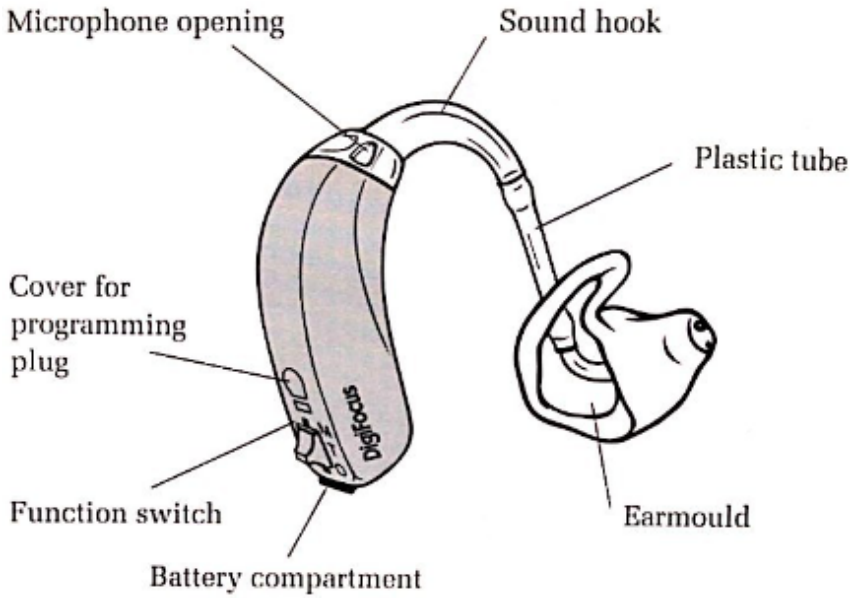
Detailed, up-to-date compatibility lists of Phonak's MLx MicroLink receiver are available online at:

www.phonak.com/mlx

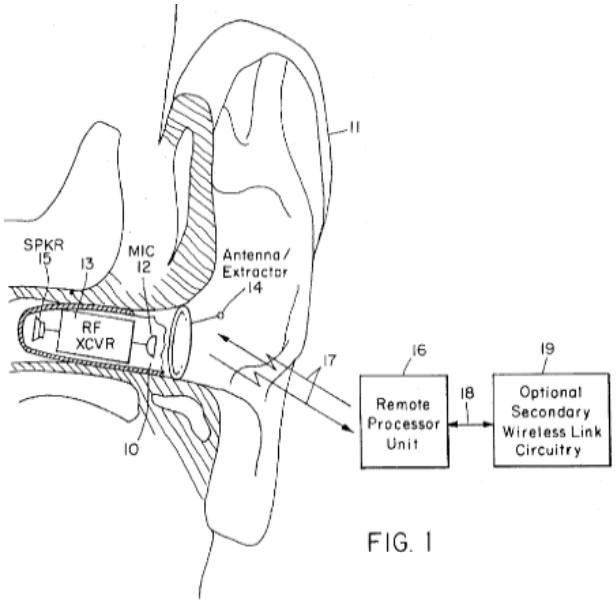
Phonak – Claro 211 Product Information at Page 2

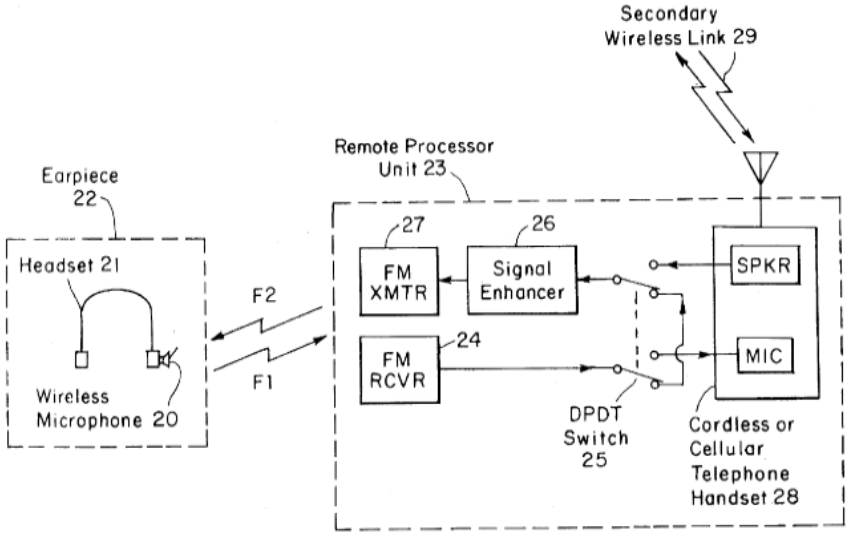
Principle of FM operation

The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi.
The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument.
The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program.

Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Oticon – DigiFocus White Paper (1996) at Page 1 DigiFocus is a fully digital hearing instrument built around the digital audio processor, DAP. This processor is designed to have a specific functionality and to comply with a series of physical, acoustical and audiological demands. □

Claims	Prior Art
	Oticon – DigiFocus Description at p. 1 DigiFocus is available both as behind-the-ear models (BTE) in colours which match the hair tone, and as in-the-ear (ITE) models which fit discreetly into the ear.  DigiFocus Compact, Canal, Half Shell and Low Profile. Oticon DigiFocus – What Users Say About DigiFocus at p. 1 Since January 1996, hundreds of hearing instrument users around the world have participated in field trials of DigiFocus. They all have varying degrees of hearing loss and had worn other instruments for years. During the field trials, Oticon carefully monitored their responses, and the results confirm that DigiFocus is a revolution, not only in terms of technology, but also in improving the quality of life for people who use it. Oticon – Products Index at Page 1 DigiFocus - a 100% digital hearing instrument with fully automatic operation, DigiFocus offers the ultimate in speech understanding. Utilising seven tone band amplification to get the most out of the remaining hearing, DigiFocus is currently the most advanced hearing instrument on the market. Anderson '783 at Abstract A hearing aid or audio communication system includes an earpiece (10) that can be hidden in the ear canal, and which communicates wirelessly with a remote processor unit, or RPU (16), that enhances audio signals and can be concealed under clothing. Sounds from the environment are picked up by a microphone (12) in the earpiece and sent with other information over a two-way wireless link (17) to the RPU (16). The wireless link (17) uses microwaves for component miniaturization. Furthermore, use of radar technology to implement the wireless link (17), with an RPU (16) interrogator and earpiece (10)

Claims	Prior Art
	transponder, reduces earpiece size and power, as no microwave oscillator is needed in the earpiece (10). Optional secondary wireless link circuitry (19) can be used between the RPU (16) and a cellular telephone system or other sources of information. Electronic voice recognition and response can control system operation. Anderson '783 at FIG. 1  FIG. 1

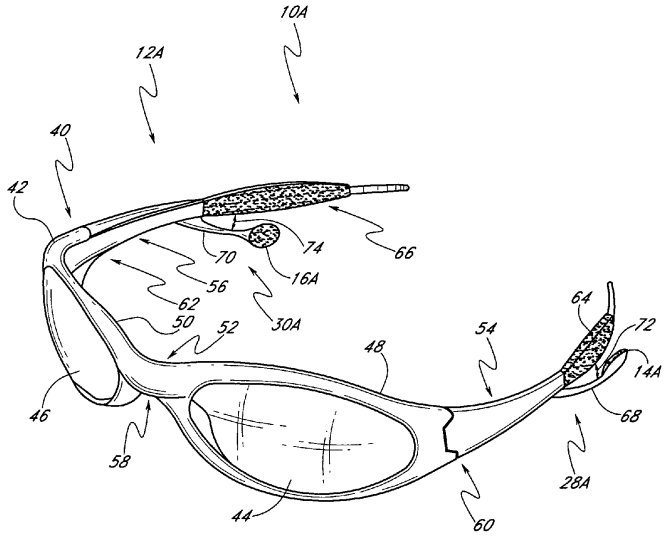
Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2 Anderson ’783 at 1:5–18 The present invention is directed to hearing aids, and in particular to a hearing aid having an earpiece housing worn in or at the ear and a remote processor unit (RPU) worn by or located near the user that wirelessly receives signals from and transmits signals to the earpiece. The present invention is also directed to the use of hearing aid systems as wireless communication devices for hands-free cellular telephone and mobile radio communication "handsets," covert operation and control of hearing aids, hearing protection and noise cancellation simultaneous with binaural hearing aid functions, hearing test equipment, location of misplaced hearing aid system components and wireless cochlear implants. Anderson ’783 at 1:50–65 The present invention uses an audio path that is different from the audio path used by devices in the prior art. The present invention uses a primary two-way wireless link (not a one-way link as in the prior art) between an earpiece worn in or at the user's ear and an RPU worn by or located near the user. In the present invention, audio signals from the environment are picked up by a microphone in the earpiece (not a hand-held microphone as used by wireless hearing aids in the prior art) and transmitted over the primary two-way wireless link to the RPU (instead of going to processor electronics contained in the earpiece, as

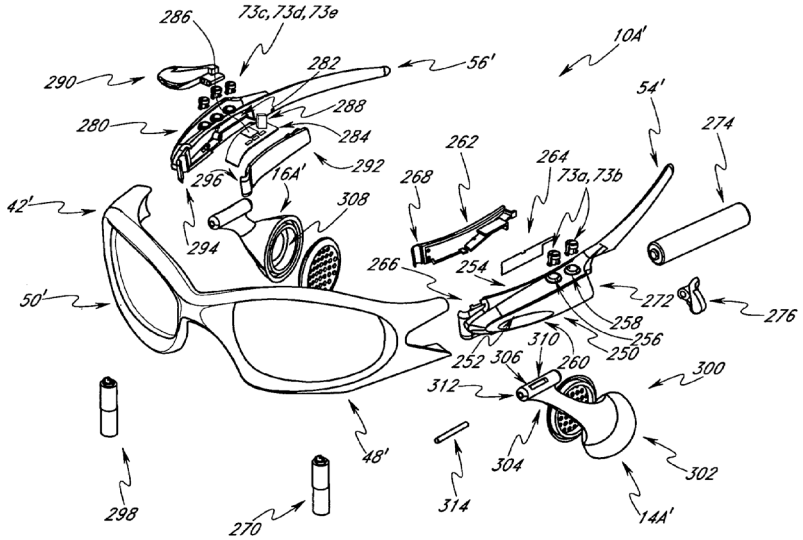
Claims	Prior Art
	in basic hearing aids known in the prior art), where the audio signals are enhanced according to the user's needs before transmission over the primary wireless link to the earpiece. Signal processing is performed in the RPU rather than the earpiece to take advantage of relaxed size and power constraints. Anderson '783 at 3:13–18 FIG. 1 is a block diagram system overview illustrating how an earpiece (comprising a microphone, RF transceiver, dual-use antenna/extractor and speaker) is worn in the ear, communicates via a primary two-way RF link with an RPU, and also communicates via an optional secondary wireless link to a telephone system. Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the

Claims	Prior Art
	DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired.

Claims	Prior Art
	Anderson ’783 at 27:4–24 Hearing test: The present invention allows hearing tests to be performed via the earpiece normally worn by the user, without the need for additional expensive test equipment. All signal generation capabilities necessary to perform hearing tests are available in the hearing aid system comprising an earpiece 10 and RPU 16. Specifically, the RPU DSP 948 can be used to generate tones of varying frequencies and amplitudes, as well as other signals known to be useful for hearing test purposes (e.g., nonsense syllables) that are converted to acoustic waves by the earpiece. Such tones are used to perform a hearing test that determines appropriate gain vs. frequency parameters for a program stored in the RPU DSP 948 that performs signal enhancement to compensate for the user's hearing loss. Note that the hearing test program that controls the RPU DSP 948 during the hearing test can be temporarily stored in the RPU DSP 948 for the duration of the test, then deleted upon completion of the test to allow re-use of RPU DSP 948 memory resources during normal operation. The hearing test program may be loaded into the RPU DSP 948 through the secondary wireless link 944 or a wired peripheral link 950. Enzmann ’730 at Abstract An aided ear bud device is provided that operates as a hearing aid device as well as an ear bud device for a wireless phone. The aided ear bud normally operates as a hearing aid device. The aided ear bud state is changed by a verified input from a wireless phone. Upon the change in state, the microphone input from the aided ear bud is remotely output to the wireless phone while the incoming transmission from the wireless phone is combined with the microphone input and output to the user. The ear bud device can be modified to receive analog or digital inputs from other outside sources. Enzmann ’730 at 1:63–66 Current ear bud devices must be placed in or in close proximity to the ear, meaning that the ear bud placement conflicts with the placement of the hearing aid. Enzmann ’730 at 2:17–23 Therefore, this invention provides systems and methods for adapting a hearing aid to operate as an ear bud device in addition to providing conventional hearing aid functions. Using this method, this invention is able to address many of the issues hearing impaired wireless phone users encounter while addressing the inherent problems associated with the conventional solutions.

Claims	Prior Art
	Enzmann ’730 at 3:11–14 The exemplary embodiments of the systems and methods according to this invention provide a method for adapting a hearing aid so that the hearing aid is able to function as an ear bud device i.e., an aided ear bud. Enzmann ’730 at 3:35–40 The manipulating processes of the DSP 120 serves both to amplify the converted signal from the A/D converter 110 and correct any degraded signal areas so that an improved signal is sent to the speaker 140. The manipulation processes may also adjust the incoming signal based on a preprogrammed frequency mapping of the user's hearing characteristics. Enzmann ’730 at 5:20–30 The signal 226b passed to the op amp 223 is representative of ambient noise in the aided ear bud 200. This signal 226b is typically reduced in volume and combined with the input from the signal-processing function 225 to produce a mixed signal. The signal processing function 225 processes signal 228 from the TX/RX unit 260 and provides a second processing step so that the timing of the incoming signals from the microphone and the wireless phone are in sync and can be effectively combined to create a true representation of sound to the user. The op amp 223 outputs the mixed signal to the D/A converter 240, and finally to the speaker 250. Enzmann ’730 at 5:34–40 The signal 239, output to the speaker 250, is a combined signal based on inputs from the microphone 205 and the wireless phone 270. The combined signal is made to duplicate the experience of a non-hearing impaired person, in that the non-hearing impaired person hears a combination of his or her own voice and ambient noise when speaking into a wireless phone.

Claims	Prior Art
	Jannard ’526 at FIG. 3A  *FIG. 3A*

Claims	Prior Art
	Jannard ’526 at FIG. 3J  *FIG. 3J* Jannard ’526 at 1:20–24 The present invention is directed to wearable audio devices, and in particular, devices that humans can wear on their head and which include audio electronics such as, for example, speakers, microphones, and/or interface electronics for interacting with a wireless network. Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted telecommunication signal.

Claims	Prior Art
	Jannard ’526 at 5:9–14 Advantageously, the support 12 is configured to support the speakers 14, 16 at a position juxtaposed to the ears 20, 22, respectively, without applying a force against the ears 20, 22 sufficient for anchoring the speakers 14, 16 in place. Thus, the support 12 contacts the head 18 at a position other than the outer surface of the ears 20, 22. Jannard ’526 at 5:43–46 As noted above, the speakers 14, 16 are spaced from the ears 20, 22 such that the speakers 14, 16 do not engage the outer surface of the ears 20, 22 with sufficient force to provide an anchoring effect for the speakers 14, 16. Jannard ’526 at 8:38–42 The audio device 10A can be used as an audio output device for any type of device which provides an audio output signal. The audio device 10A can include an audio input terminal disposed anywhere on the eyeglass 40 for receiving a digital or analog audio signal. Jannard ’526 at 19:41–45 As illustrated in FIG. 8, the audio device 10C can be worn on the head 18 of a user U. Preferably, the audio device 10C is configured to provide one or two-way wireless communication with a source device, or the source device can be incorporated into the audio device 10C. Jannard ’526 at 19:48–54 The user U can carry a “body borne” source device B such as, for example, but without limitation, a cellular phone, an MP3 player, a “two-way” radio, a palmtop computer, or a laptop computer. As such, the user U can use the audio device 10C to receive and listen to audio signals from the source device B, and/or transmit audio signals to the source device B. Jannard ’526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E.

Claims	Prior Art
	Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. To the extent that the DHA systems not disclose any elements of this claim limitation or specific details of implementation, a POSITA would have been motivated to combine elements of DHA system with features known to them and/or disclosed in secondary references, including Anderson or Enzmann or Jannard, to meet the claim limitations with reasonable expectation of success. A POSITA would have understood prior art DHA systems to include features and functionality relating to, for example, a personal audio system that generates customized output. It would have been obvious to a POSITA to combine Anderson’s teachings for at least the reason that it provides additional information to fill in the missing details of well–known digital hearing aid components and/or functionality. Moreover, such references are readily combinable with DHA systems. For example, the Phonak system, and secondary references like Anderson, Enzmann, and Jannard share various structural and functional similarities. For example, each discloses ear-worn devices for providing audio output to a user, and include similar components, such as directional microphones, speakers, wireless transceivers, control circuitry, a power source for providing power to the device, and altering audio

Claims	Prior Art
	output. Accordingly, combining the teachings of the Phonak system and Anderson, Enzmann, and/or Jannard would involve combining known components to yield predictable results. To the extent that DHA systems and/or Anderson or Enzmann or Jannard do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1a] at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user;	DHA systems, alone or in view of Anderson, Enzmann or Jannard, expressly or inherently disclose "at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user." Phonak – Perseo Brochure at p. 8 

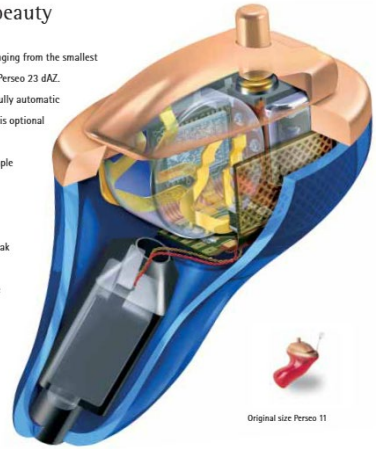
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.

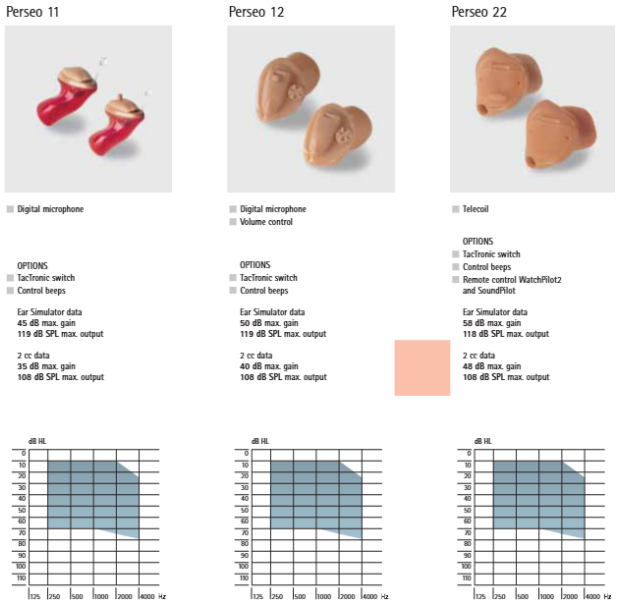


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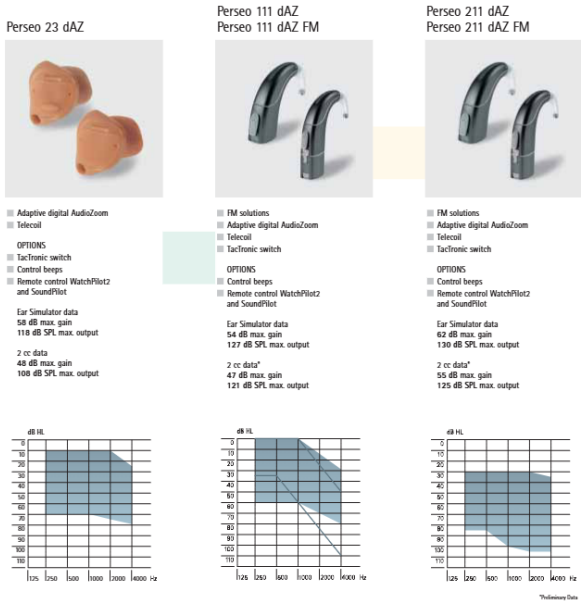
IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 34

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 26



Claims	Prior Art
	Phonak – Perseo Brochure at Page 27



IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 37

Claims	Prior Art
	Phonak – Perseo Brochure at Page 28



Phonak - SmartLink SX User Guide at p. 32



Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 398 Differences in performance between omnidirectional and directional microphones were evaluated between two loudspeaker conditions (single loudspeaker at 180°; diffuse using eight loudspeakers set 45° apart) and two types of noise (steady-state HINT noise; R-Space™ restaurant noise). Twenty-five participants were fit bilaterally with Phonak Perseo hearing aids using the manufacturer's recommended procedure. After wearing the hearing aids for one week, the Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

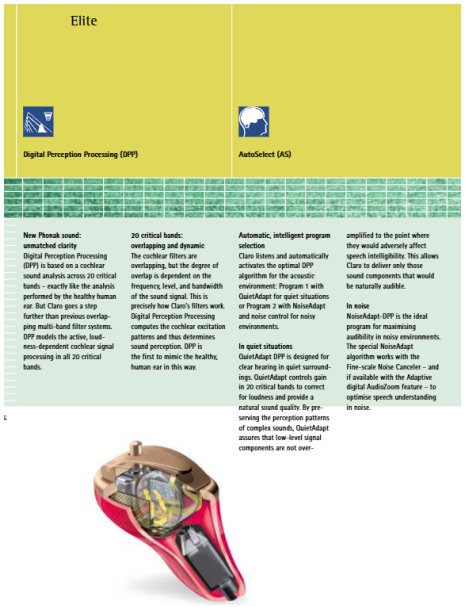
Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

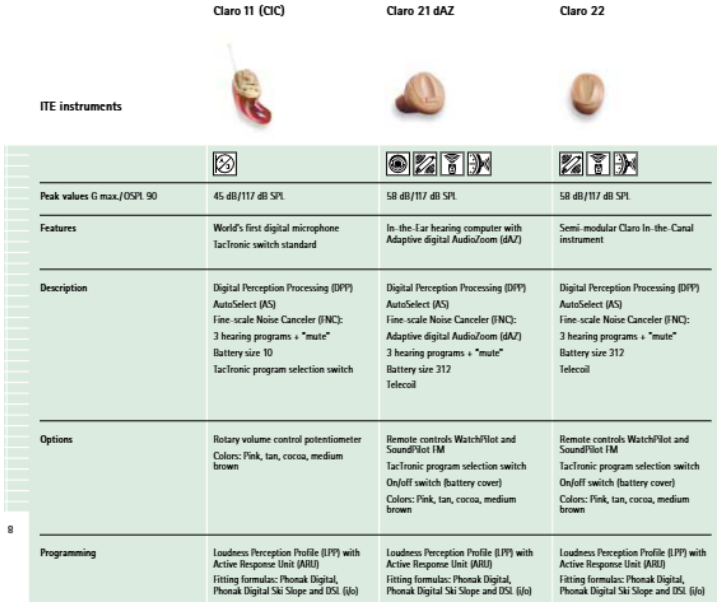
Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 6

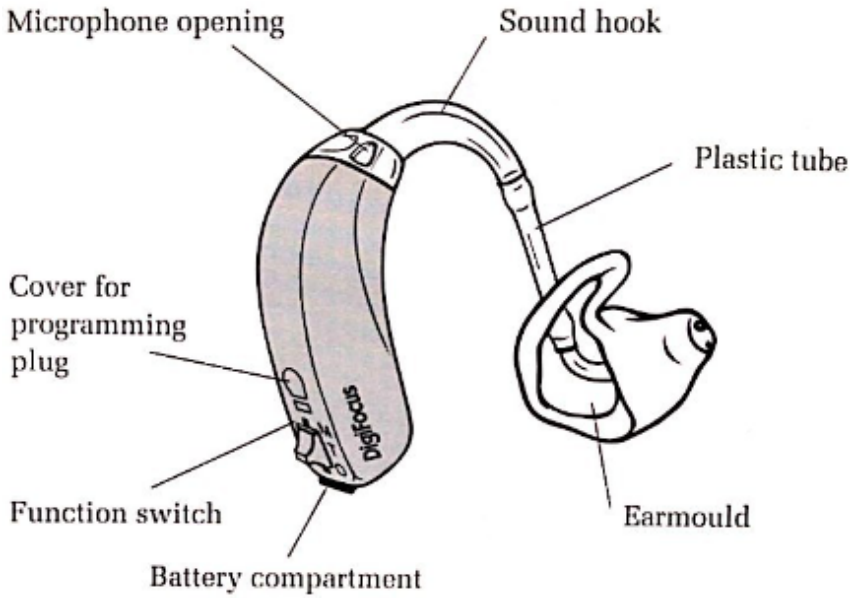


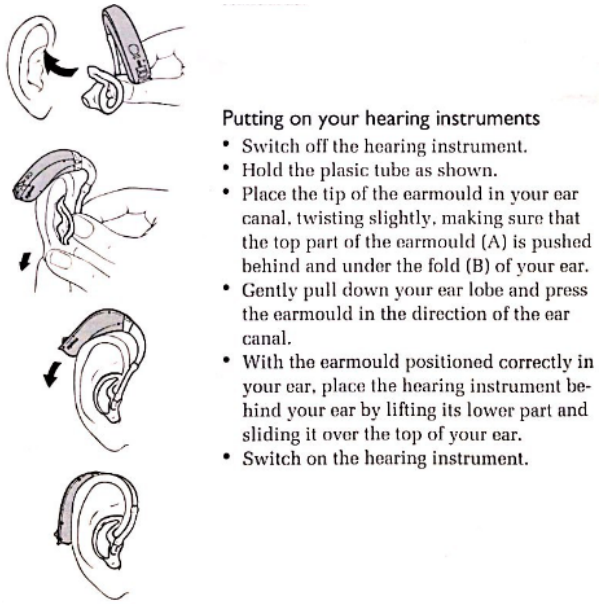
Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 8

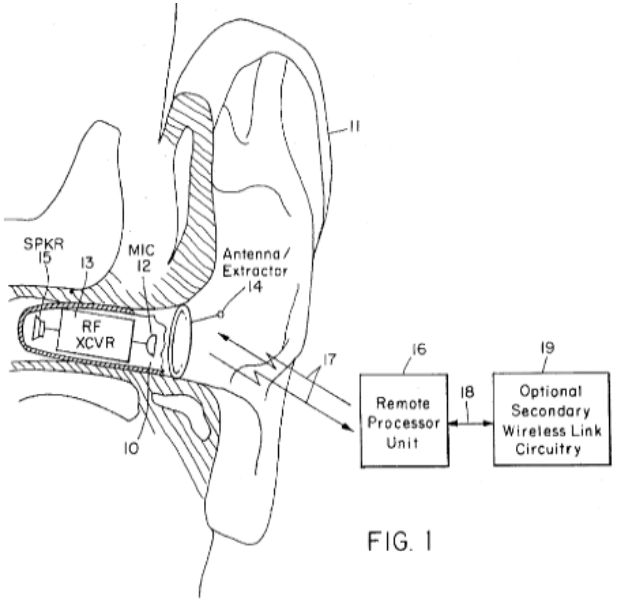


	Claro 11 (CIC)	Claro 21 dAZ	Claro 22
ITE instruments			
Peak values G max./OSPL 90	45 dB/117 dB SPL	58 dB/117 dB SPL	58 dB/117 dB SPL
Features	World's first digital microphone TacTronic switch standard	In-the-Ear hearing computer with Adaptive digital AudioZoom (dAZ)	Semi-modular Claro In-the-Canal instrument
Description	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC): 3 hearing programs + "mute" Battery size 10 TacTronic program selection switch	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC): Adaptive digital AudioZoom (dAZ) 3 hearing programs + "mute" Battery size 312 Telecoil	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC): 3 hearing programs + "mute" Battery size 312 Telecoil
Options	Rotary volume control potentiometer Colors: Pink, tan, cocoa, medium brown	Remote controls WatchPilot and SoundPilot FM TacTronic program selection switch On/off switch (battery cover) Colors: Pink, tan, cocoa, medium brown	Remote controls WatchPilot and SoundPilot FM TacTronic program selection switch On/off switch (battery cover) Colors: Pink, tan, cocoa, medium brown
Programming	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)

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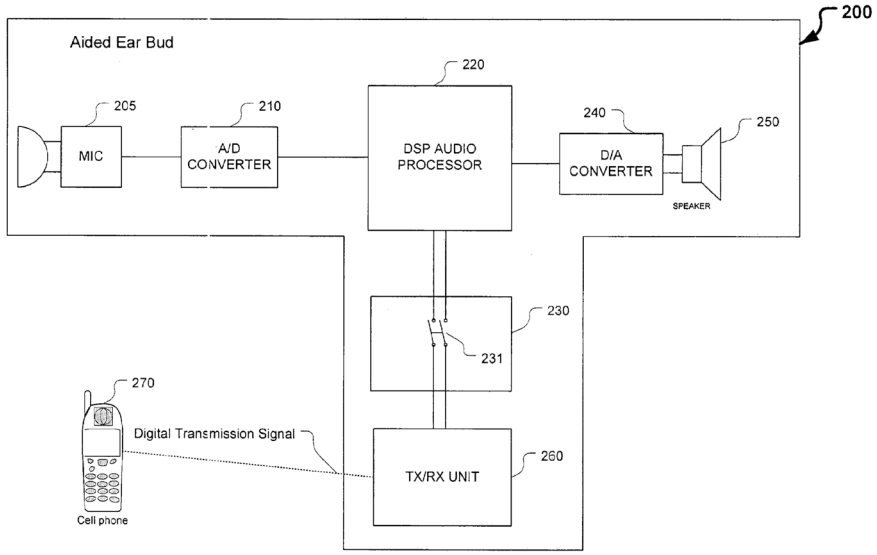
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4 

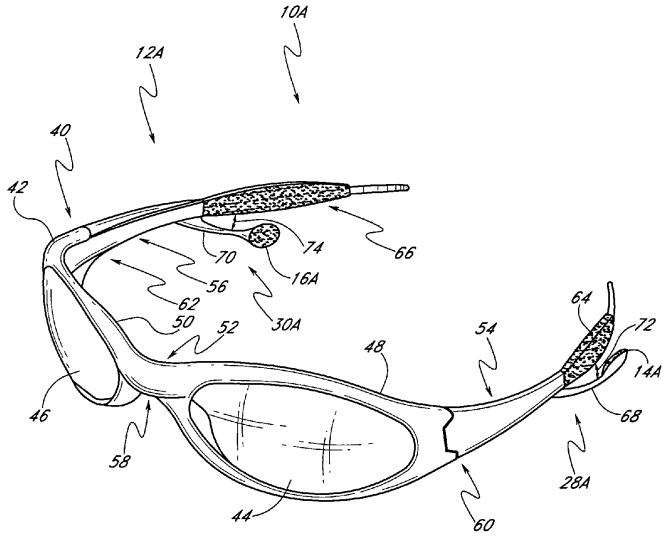
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at p. 6  Putting on your hearing instruments • Switch off the hearing instrument. • Hold the plasic tube as shown. • Place the tip of the earmould in your ear canal, twisting slightly, making sure that the top part of the earmould (A) is pushed behind and under the fold (B) of your ear. • Gently pull down your ear lobe and press the earmould in the direction of the ear canal. • With the earmould positioned correctly in your ear, place the hearing instrument behind your ear by lifting its lower part and sliding it over the top of your ear. • Switch on the hearing instrument. Oticon – DigiFocus White Paper (1996) at Page 11 Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time.

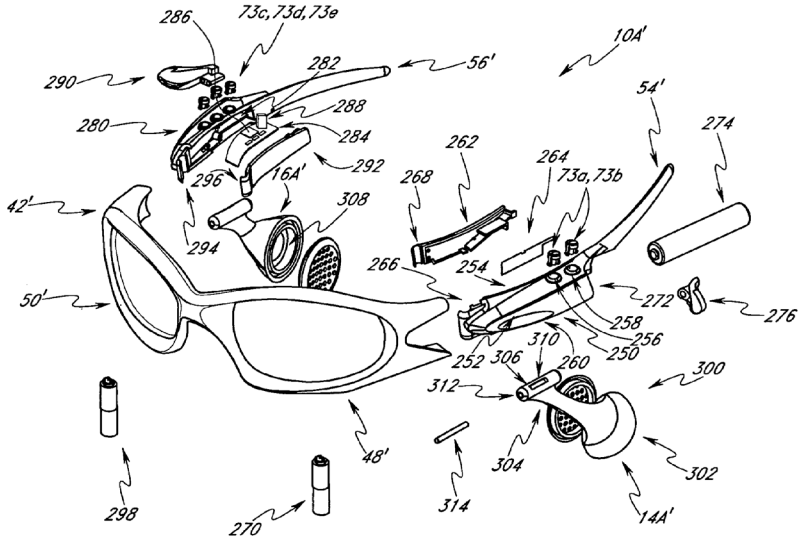
Claims	Prior Art
	Oticon – DigiFocus Description at p. 1 DigiFocus is available both as behind-the-ear models (BTE) in colours which match the hair tone, and as in-the-ear (ITE) models which fit discreetly into the ear.  DigiFocus Compact, Canal, Half Shell and Low Profile. Anderson '783 at FIG. 1  FIG. 1 Anderson '783 at 3:13–18 FIG. 1 is a block diagram system overview illustrating how an earpiece (comprising a microphone, RF transceiver, dual-use antenna/extractor and speaker) is worn in the ear, communicates via a primary two-

Claims	Prior Art
	way RF link with an RPU, and also communicates via an optional secondary wireless link to a telephone system. Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 4:30–33 Although the system described here is monaural, it will be clear to those skilled in the art that a binaural (stereo) system can be constructed based on the same principles as the monaural system. Anderson '783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities

Claims	Prior Art
	of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired.

Claims	Prior Art
	Enzmann ’730 at FIG. 2  FIG. 2 Enzmann ’730 at 3:61–67 The aided ear bud 200 comprises a microphone 205, an A/D converter 210, a digital signal audio processor (DSP) 220, a D/A converter 240 and a speaker 250. However, in addition to those items, the aided ear bud adds a source control module 230 and a transmit/receive (TX/RX) unit 260 to provide the necessary functions of this invention. Enzmann ’730 at 4:26–29 Ambient noise and the user's voice are captures by the microphone 205 and processed and combined with signals from the TX/RX unit 260 in the DSP 220 for output through the D/A converter 246 and the speaker 250. Enzmann ’730 at 8:3–7 At step S2000, the digital signal is sent to the DSP to be processed. The process then continues to step S2100, where the corrected signal is output to the speaker of the ear bud. The speaker of the ear bud is typically positioned in or in close proximity to the user's ear.

Claims	Prior Art
	Jannard ’526 at FIG. 3A  *FIG. 3A*

Claims	Prior Art
	Jannard ’526 at FIG. 3J  *FIG. 3J* Jannard ’526 at 1:34–37 The audio output for many of these systems is typically directed to the wearer through the use of transducers physically positioned in or covering the ear, such as earphones and headphones. Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted telecommunication signal.

Claims	Prior Art
	Jannard ’526 at 2:26–34 In accordance with another embodiment of at least one of the inventions disclosed herein, an eyeglass frame comprises a support for supporting at least one lens in the path of a wearer's field of view, a first ear stem attached to the support, a second ear stem attached to the support, and at least one microphone supported by at least one of the support, first ear stem, and second ear stem. The microphone is advantageously arranged to face towards the head of a wearer of the eyeglass frame. Jannard ’526 at 5:9–14 Advantageously, the support 12 is configured to support the speakers 14, 16 at a position juxtaposed to the ears 20, 22, respectively, without applying a force against the ears 20, 22 sufficient for anchoring the speakers 14, 16 in place. Thus, the support 12 contacts the head 18 at a position other than the outer surface of the ears 20, 22. Jannard ’526 at 5:43–46 As noted above, the speakers 14, 16 are spaced from the ears 20, 22 such that the speakers 14, 16 do not engage the outer surface of the ears 20, 22 with sufficient force to provide an anchoring effect for the speakers 14, 16. Jannard ’526 at 7:27–31 In the illustrated embodiment, the support members 28A, 30A are in the form of support arms 68, 70 extending downwardly from the ear stems 54, 56, respectively. As such, the speakers 14A, 16A can be precisely positioned relative to the ears 20, 22 (FIG. 1) of a wearer's head 18. Jannard ’526 at 8:18–23 This construction provides a further advantage in that if a user rotates the eyeglass 40 such that the lenses 44, 46 are moved upwardly out of the field of view of the wearer, the speakers 14A, 16A can be more easily maintained in alignment with the ears 20, 22 of the wearer. Jannard ’526 at 10:51–57 Further, some wearers of the audio device 10A′ may prefer to press the speakers 14A′, 16A′ into contact with the outer surfaces of their ears. For example, some users may desire to experience to loudest possible volume from the speakers 14A′, 16A′. Thus, by pressing the speakers 14A′, 16A′ against their ears, the

Claims	Prior Art
	perceived volume of the sound emitted from the speakers 14A′, 16A′ will be the greatest. Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. To the extent that the DHA systems not disclose any elements of this claim limitation or specific details of implementation, a POSITA would have been motivated to combine elements of DHA system with features known to them and/or disclosed in secondary references, including Anderson, Enzmann or Jannard, to meet the claim limitations with reasonable expectation of success. A POSITA would have understood prior art DHA systems to include features and functionality relating to, for example, a pair of ear buds, each configured to be placed bilaterally, one in an ear of a user. It would have been obvious to a POSITA to combine teachings, like those of Anderson, Enzmann or Jannard), for at least the reason that it provides information to fill in the missing details of well–known digital hearing aid components and/or functionality, such as speakers, and to improve user experience for users of digital hearing aids seeking to compensate for their hearing impairment. A POSITA would have understood this to also include details relating to directional

Claims	Prior Art
	microphones, wireless transceivers, control circuitry, a power source for providing power to the device, and a mechanical switch for altering audio output. Moreover, such references are readily combinable with DHA systems. For example, the Phonak system, and secondary references like Anderson, Enzmann and Jannard, share various structural and functional similarities. For example, each discloses ear-worn devices for providing audio output to a user, and include similar components, such as directional microphones, speakers, wireless transceivers, control circuitry, a power source for providing power to the device, and altering audio output. Accordingly, combining the teachings of the Phonak system and Anderson, Enzmann, or Jannard would involve combining known components to yield predictable results. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1b] a wireless receiver configured to wirelessly receive audio output signals to be output to the user via the at least one speaker; and	DHA systems, alone or in view of Anderson, Enzmann or Jannard, expressly or inherently disclose "a wireless receiver configured to wirelessly receive audio output signals to be output to the user via the at least one speaker." Phonak – Perseo Brochure at Page 5 The Science of PersonalLogic™ Phonak has applied knowledge from new research and past experience to add a new dimension of individualized logic to automatic hearing systems – PersonalLogic. Combining new and proven features, PersonalLogic optimizes the functioning, operation and fitting of hearing systems for each individual user. Thanks to PersonalLogic, Perseo takes customization to a level beyond "custom-fitted" related to individual hearing loss. The new digital system from Phonak is the first hearing system with customized automatic operation, as opposed to operation based on factory set, unchangeable rules. Now, Perseo gives hearing professionals control over automatic operation, allowing them to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 15

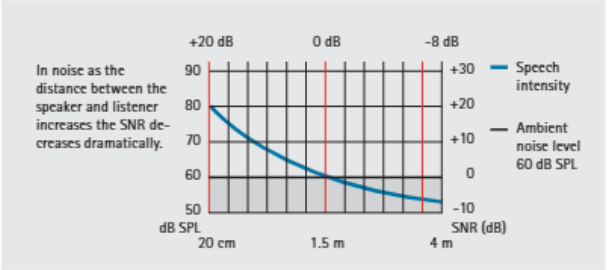
The perfectly integrated communication system

FM MicroLink

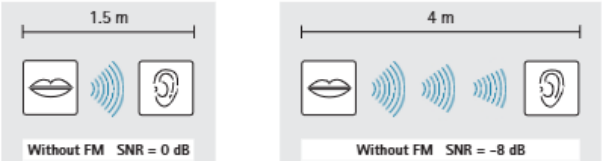
FM technology has allowed thousands of children to participate in standard education settings and to be actively involved in social activities. Phonak pioneered this development and the integration of FM into hearing instruments to produce communication systems that are effective in even the most demanding situations. Since the innovation of miniaturized Phonak **MicroLink** systems, FM has become an important solution also for adults with a high communication need in work and leisure environments.

All Perseo BTEs have been designed to incorporate MicroLink with its fully integrated appearance and operation. This integrated system is unique to Phonak.

Speech-to-Noise Ratio (SNR) as a function of distance between speaker and listener



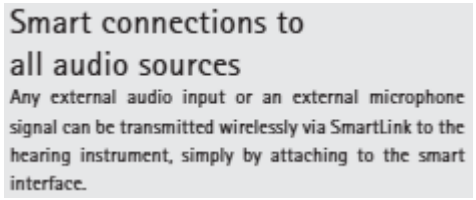
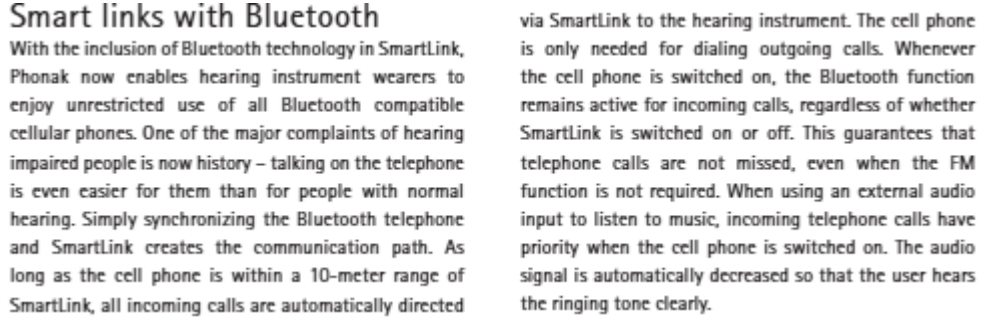
Speech-to-Noise Ratio without FM is distance dependent



IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 54

Claims	Prior Art
	Phonak – Perseo Brochure at Page 28  Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear!

Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Phonak – Online Interview re Perseo (April 7, 2003) at Page 9 **Liz Brassine/Phonak:** It is a digital microphone on the CIC. There are several advantages of a digital microphone over analog mics. First, increased dynamic range is provided. Additionally, with a digital mic, the analog-to-digital converter is housed within the microphone itself. This eliminates the wire between the AD-converter and the microphone. The wire is the source for ultra-high frequency interference, such as Bluetooth. Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink SX is the first communication device to incorporate • Advanced FM features • Hearing Instrument remote control • A Bluetooth link for cell phone use

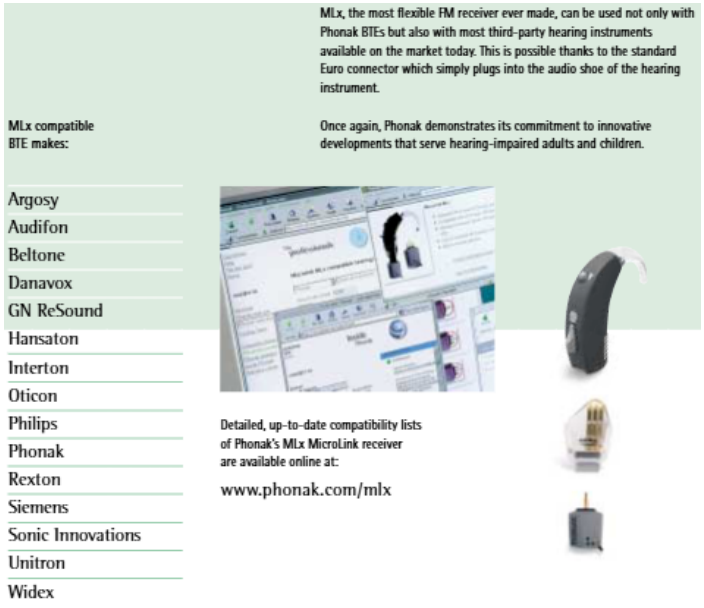
Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 4  Smart connections to all audio sources Any external audio input or an external microphone signal can be transmitted wirelessly via SmartLink to the hearing instrument, simply by attaching to the smart interface. Phonak - SmartLink SX - FM Systems Manual at p. 5  Smart FM transmission The proven Phonak FM technology clearly transmits all incoming auditory signals from SmartLink to the receiver attached to the user's hearing instrument. Phonak - SmartLink SX - FM Systems Manual at p. 10  Smart links with Bluetooth With the inclusion of Bluetooth technology in SmartLink, Phonak now enables hearing instrument wearers to enjoy unrestricted use of all Bluetooth compatible cellular phones. One of the major complaints of hearing impaired people is now history – talking on the telephone is even easier for them than for people with normal hearing. Simply synchronizing the Bluetooth telephone and SmartLink creates the communication path. As long as the cell phone is within a 10-meter range of SmartLink, all incoming calls are automatically directed via SmartLink to the hearing instrument. The cell phone is only needed for dialing outgoing calls. Whenever the cell phone is switched on, the Bluetooth function remains active for incoming calls, regardless of whether SmartLink is switched on or off. This guarantees that telephone calls are not missed, even when the FM function is not required. When using an external audio input to listen to music, incoming telephone calls have priority when the cell phone is switched on. The audio signal is automatically decreased so that the user hears the ringing tone clearly.

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 10wire  0 Phonak - SmartLink SX - FM Systems Manual at p. 11 Smart digital synthesizer technology Digital frequency synthesizer technology, only available with Phonak FM communication systems, enables online frequency synchronization and scanning of the frequencies used. Once a frequency is selected on the SmartLink, it automatically sends a signal to the FM receiver that adjusts the frequency of the receiver accordingly. This automatic synchronization process occurs every time the SmartLink is switched on. In cases where more than one FM transmitter is in use, at lectures or presentations for example, the SmartLink has an exclusive scan function. This seeks out the frequency being used by the other transmitter and sets the receiver frequency accordingly.

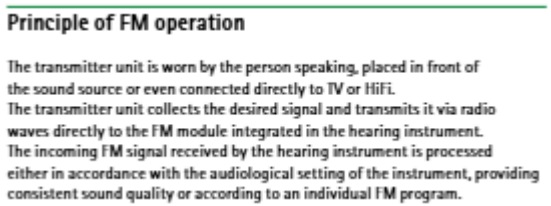
Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 14  Phonak - SmartLink SX User Guide at p. 32 

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

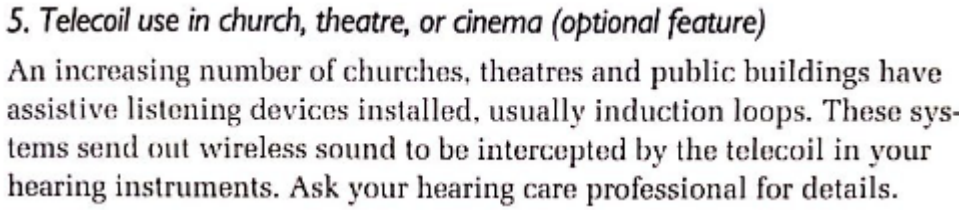
Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 30



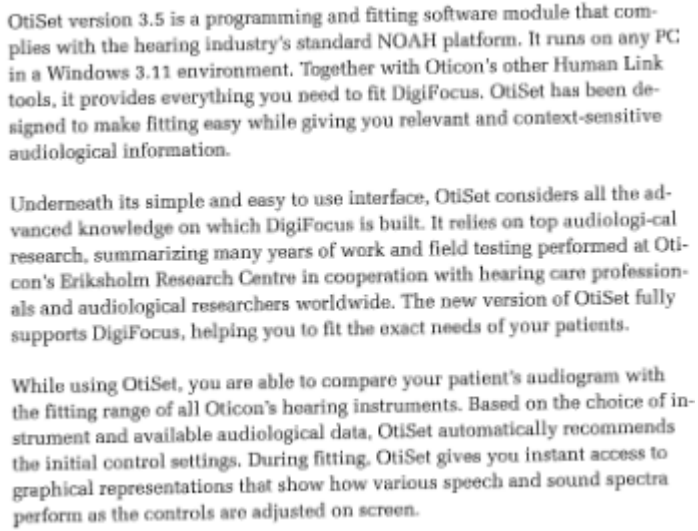
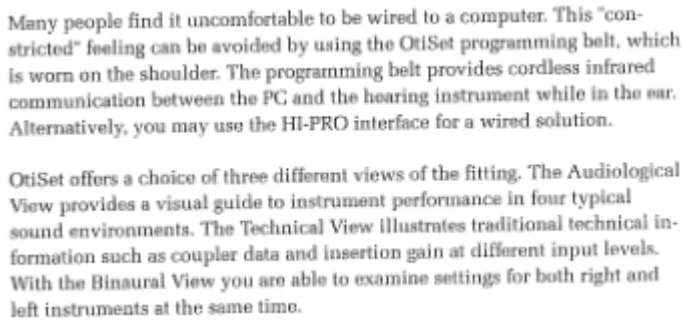
Phonak – Claro 211 Product Information at Page 2

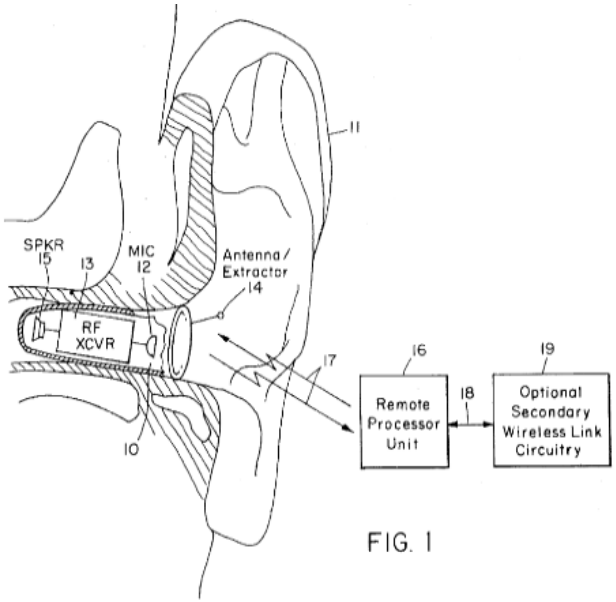


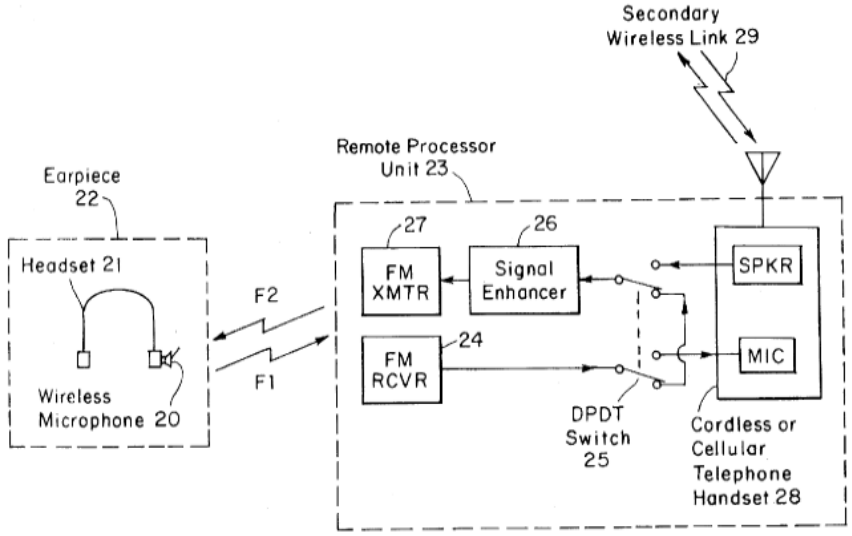
Oticon – DigiFocus Compact User Instructional Brochure at Page 11

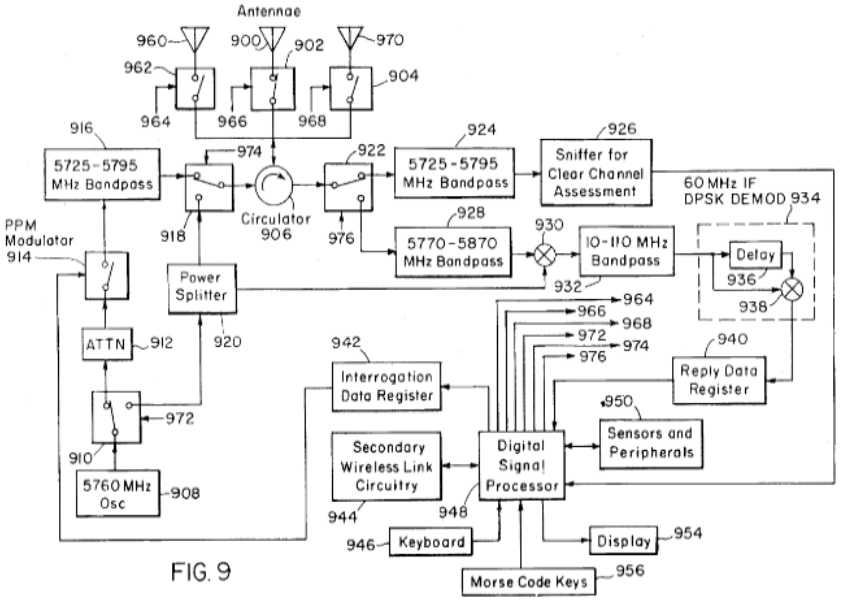


Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 1 DigiFocus is a fully digital hearing instrument built around the digital audio processor, DAP. This processor is designed to have a specific functionality and to comply with a series of physical, acoustical and audiological demands. □ Oticon – DigiFocus White Paper (1996) at Page 4 The input signal from the microphone or the telecoil is digitized and fed to a filterbank, which divides the signal into seven band-pass filters with centre frequencies at: 250, 700, 1200, 2000, 3000, 4000 and 5000 Hz. The filterbank covers the frequency range from 125 to about 6000 Hz. The output from each band-pass filter is equipped with an attenuator by which each filtered signal can be weighted. This allows for a very precise fitting in accordance with the audiometric profile of the hearing impaired. Oticon – DigiFocus White Paper (1996) at Page 9 With DigiFocus you have an extremely flexible hearing instrument. By using the right tools you can define a personal, social and audiological profile for each individual patient. This enables you to utilize the capacity of DigiFocus to the fullest extent. The Human Link tools for DigiFocus include: Oticon – DigiFocus White Paper (1996) at Page 9 • The DigiFocus diary enables you to determine quickly when a hearing instrument needs adjustment • The fine-tuning recommendations help you to adjust the settings • OtiSet, a PC based software program, and the programming belt enable you to fit and fine tune DigiFocus while the client is wearing the instrument. This takes place without your patient being wired to the PC.

Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 1-0  OtiSet version 3.5 is a programming and fitting software module that complies with the hearing industry's standard NOAH platform. It runs on any PC in a Windows 3.11 environment. Together with Oticon's other Human Link tools, it provides everything you need to fit DigiFocus. OtiSet has been designed to make fitting easy while giving you relevant and context-sensitive audiological information. Underneath its simple and easy to use interface, OtiSet considers all the advanced knowledge on which DigiFocus is built. It relies on top audiologi-cal research, summarizing many years of work and field testing performed at Oticon's Eriksholm Research Centre in cooperation with hearing care professionals and audiological researchers worldwide. The new version of OtiSet fully supports DigiFocus, helping you to fit the exact needs of your patients. While using OtiSet, you are able to compare your patient's audiogram with the fitting range of all Oticon's hearing instruments. Based on the choice of instrument and available audiological data, OtiSet automatically recommends the initial control settings. During fitting, OtiSet gives you instant access to graphical representations that show how various speech and sound spectra perform as the controls are adjusted on screen. Oticon – DigiFocus White Paper (1996) at Page 11  Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time. Anderson ’783 at Abstract A hearing aid or audio communication system includes an earpiece (10) that can be hidden in the ear canal, and which communicates wirelessly with a remote processor unit, or RPU (16), that enhances audio signals and can be concealed under clothing. Sounds from the environment are picked up by a microphone (12) in the earpiece and sent with other information over a two-way wireless link (17) to the RPU (16). The wireless link (17) uses microwaves for component miniaturization. Furthermore, use of radar technology to implement the wireless link (17), with an RPU (16) interrogator and earpiece (10) transponder, reduces earpiece size and power, as no microwave oscillator is needed in the earpiece (10).

Claims	Prior Art
	Optional secondary wireless link circuitry (19) can be used between the RPU (16) and a cellular telephone system or other sources of information. Electronic voice recognition and response can control system operation. Anderson ’783 at FIG. 1  FIG. 1

Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2

Claims	Prior Art
	Anderson ’783 at FIG. 9  FIG. 9 Anderson ’783 at 1:5–18 The present invention is directed to hearing aids, and in particular to a hearing aid having an earpiece housing worn in or at the ear and a remote processor unit (RPU) worn by or located near the user that wirelessly receives signals from and transmits signals to the earpiece. The present invention is also directed to the use of hearing aid systems as wireless communication devices for hands-free cellular telephone and mobile radio communication "handsets," covert operation and control of hearing aids, hearing protection and noise cancellation simultaneous with binaural hearing aid functions, hearing test equipment, location of misplaced hearing aid system components and wireless cochlear implants. Anderson ’783 at 1:50–65 The present invention uses an audio path that is different from the audio path used by devices in the prior art. The present invention uses a primary two-way wireless link (not a one-way link as in the prior art) between an earpiece worn in or at the user's ear and an RPU worn by or located near the user. In the present invention, audio signals from the environment are picked up by a microphone in the earpiece (not a hand-held microphone as used by wireless hearing aids in the prior art) and transmitted over the primary two-way wireless link to the RPU (instead of going to processor electronics contained in the earpiece, as

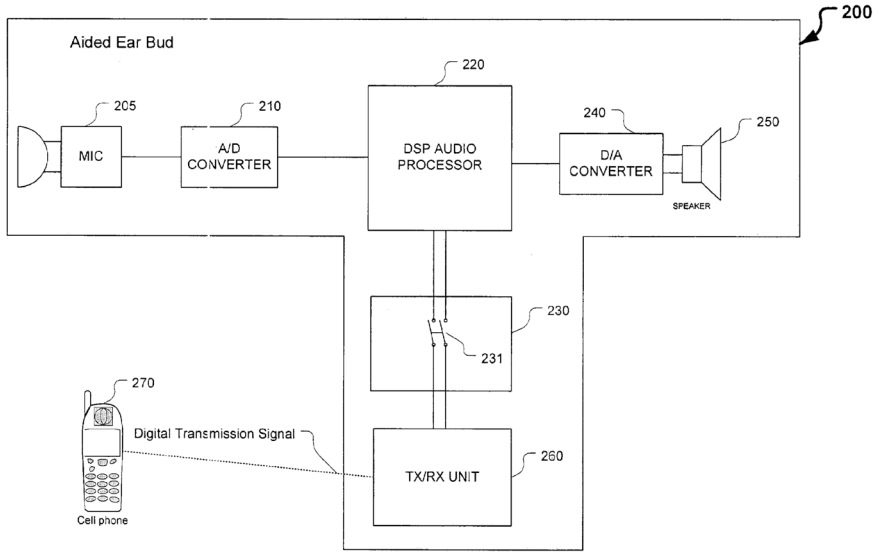
Claims	Prior Art
	in basic hearing aids known in the prior art), where the audio signals are enhanced according to the user's needs before transmission over the primary wireless link to the earpiece. Signal processing is performed in the RPU rather than the earpiece to take advantage of relaxed size and power constraints. Anderson ’783 at 2:20–39 It is an object of the present invention to provide a new and useful auditory aid for hearing-impaired persons (i.e., those having certain residual hearing) by removing audio signal enhancement functions from the earpiece and placing them in an RPU. Use of an RPU provides several advantages over systems that attempt to place all system capabilities within the earpiece. The RPU approach allows a simple earpiece design comprising a miniature low-power wireless transceiver, microphone and speaker. Note that the speaker is also known in the hearing aid art as a "receiver," but the term "speaker" is used here to avoid confusion (similarly, the term "talker," not "speaker," is used to describe a human producing vocal sounds). The resulting earpiece is extremely small, can be hidden from view in the ear canal if desired, and allows complete freedom of movement when a primary two-way wireless link to the RPU is used. Processing for all major system capabilities, such as amplification and other forms of signal enhancement, takes place in the RPU where size and power constraints are relaxed, leading to a cost-effective design. Anderson ’783 at 3:13–18 FIG. 1 is a block diagram system overview illustrating how an earpiece (comprising a microphone, RF transceiver, dual-use antenna/extractor and speaker) is worn in the ear, communicates via a primary two-way RF link with an RPU, and also communicates via an optional secondary wireless link to a telephone system. Anderson ’783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-

Claims	Prior Art
	way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 4:30–33 Although the system described here is monaural, it will be clear to those skilled in the art that a binaural (stereo) system can be constructed based on the same principles as the monaural system. Anderson '783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson '783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to

Claims	Prior Art
	a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 12:20–32 During the reply interval 52, a modulated (via modulator 830) low-power 60 MHz oscillator 835 is applied to voltage point V1, causing the diode 43 to turn on and off at a 60 MHz rate and act as a mixer. The mixer diode 43 generates frequency components centered at the difference frequency of 5700 MHz and the sum frequency of 5820 MHz. The 5820 MHz sum component passes through the resonator 42, and forms the reply which is wirelessly transmitted from the earpiece transponder antenna 40 to the RPU reply processor circuitry. The earpiece 60 MHz oscillator 835, which is only turned on during the reply interval, changes phase at specified points in time to generate the DPSK reply data. Anderson ’783 at 12:47–13:22 Remote processor unit interrogator and reply processor: The RPU, shown in FIG. 9, may use any of a wide variety of components well known in the art. The DSP 948 may contain several integrated circuits, e.g., the Motorola DSP56L002 single-chip digital signal processor, a Motorola DSP56200 cascadable-adaptive digital filter chip and supporting memory devices well known in the art. Miniature components such as dielectric resonance oscillators are available for the microwave oscillator 908, and the DPSK replies can be decoded using surface acoustic wave (SAW) or glass bulk acoustic wave (BAW) delay lines 936 in a standard delay-and-multiply demodulator circuit 934. An alternative demodulator that achieves superior noise immunity can be implemented using a phase-locked loop in a squaring loop configuration for carrier recovery instead of delay-and-multiply circuitry 934. Decoding of a DPSK waveform using such coherent detection techniques is known in the art as differentially encoded coherent phase shift keying, or DECPSK. Amplifiers using low noise pseudomorphic high electron mobility transistors (e.g., the HP ATF-36077) may be used as needed (e.g., between the circulator 906 and routing switch 922), and such amplifiers are not individually identified in FIG. 9. All switches shown in FIG. 9 are RF switches that can be implemented using, e.g., PIN diodes. Switch control connections from the DSP 948 to the RF switches are shown in FIG. 9 for clarity, and all switches remain set as shown until changed as noted. The system shown in FIG. 9 demodulates a received reply when the RF reply travels through a selected antenna 900, antenna selection switch 902, circulator 906, routing switch 922 (set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 976), bandpass filter 928 and IF mixer 930. The IF mixer 930 downconverts the reply by mixing it with a small amount of RF power from the 5760 MHz oscillator 908 via switch 910 (which is set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 972) and power

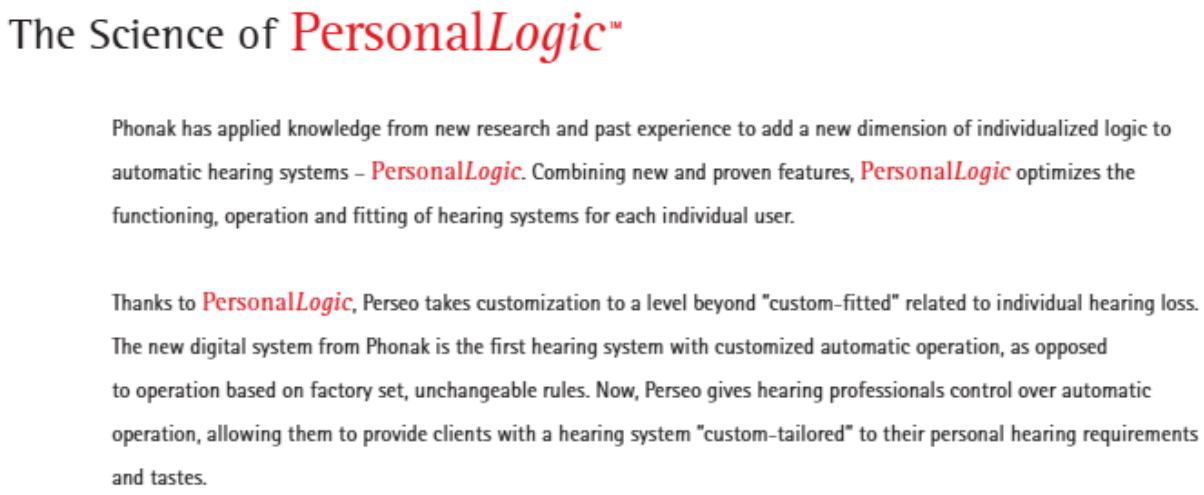
Claims	Prior Art
	splitter 920. The resulting DPSK reply has a center frequency of 60 MHz, a bandwidth of 100 MHz, and is passed through the bandpass filter 932 to the DPSK demodulator 934 which, in this example, consists of a 50 nsec delay line 936 and multiplier 938. The resulting bits are stored in the reply data register 940 for processing by the DSP 948. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 24:47–66 Morse code keys 956 or voice recognition capabilities implemented using the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. In this respect, the system acts as a "virtual handset" communication aid that replaces a conventional telephone handset in many applications

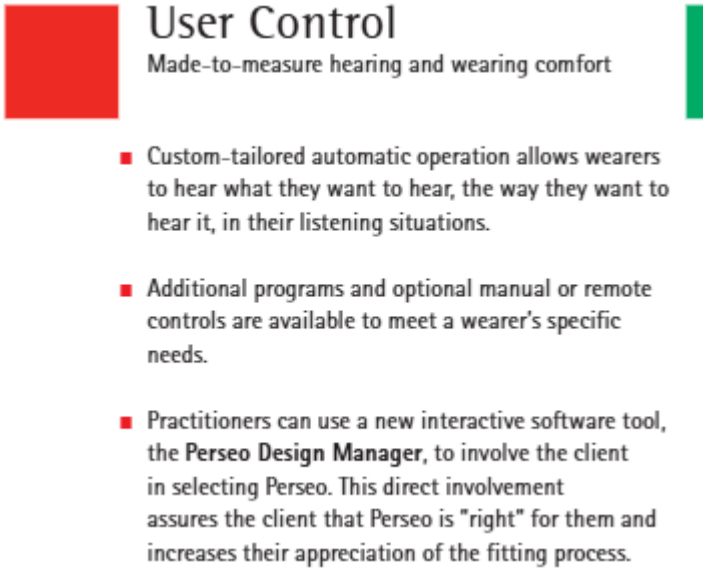
Claims	Prior Art
	(although users may additionally communicate with a conventional handset or speakerphone in the usual fashion). To accomplish the virtual handset function, the RPU 16 is equipped with optional secondary two-way wireless link circuitry 19 to communicate with a telephone or other communication network. The secondary wireless link is separate from, and does not interfere with, the primary wireless link 17 between the earpiece 10 and RPU 16. In applications where relatively high-power transmissions take place on the secondary link and the secondary link transmitter is integrated into the RPU, the RPU can be placed on a user's wrist or ankle away from vital organs to reduce any potential health risk relative to conventional wireless handsets. Anderson ’783 at 27:4–24 Hearing test: The present invention allows hearing tests to be performed via the earpiece normally worn by the user, without the need for additional expensive test equipment. All signal generation capabilities necessary to perform hearing tests are available in the hearing aid system comprising an earpiece 10 and RPU 16. Specifically, the RPU DSP 948 can be used to generate tones of varying frequencies and amplitudes, as well as other signals known to be useful for hearing test purposes (e.g., nonsense syllables) that are converted to acoustic waves by the earpiece. Such tones are used to perform a hearing test that determines appropriate gain vs. frequency parameters for a program stored in the RPU DSP 948 that performs signal enhancement to compensate for the user's hearing loss. Note that the hearing test program that controls the RPU DSP 948 during the hearing test can be temporarily stored in the RPU DSP 948 for the duration of the test, then deleted upon completion of the test to allow re-use of RPU DSP 948 memory resources during normal operation. The hearing test program may be loaded into the RPU DSP 948 through the secondary wireless link 944 or a wired peripheral link 950.

Claims	Prior Art
	Enzmann ’730 at FIG. 2  FIG. 2 Enzmann ’730 at 3:46–49 The embodiments of the present invention allow the user to hear the audio transmissions of the wireless phone in the speaker of the aided ear bud. This is achieved by transmitting the audio output to the aided ear bud. Enzmann ’730 at 4:14–16 In the second operating mode, the TX/RX unit 260 transmits and receives communications from the wireless phone 270. Enzmann ’730 at 4:26–29 Ambient noise and the user's voice are captures by the microphone 205 and processed and combined with signals from the TX/RX unit 260 in the DSP 220 for output through the D/A converter 246 and the speaker 250. Enzmann ’730 at 4:43–50 The connection shown between the TX/RX unit 260 and the wireless phone 270 is preferably a wireless

Claims	Prior Art
	digital connection, such as Bluetooth, etc. However, it should be appreciated that the connection between the TX/RX unit 260 and the wireless phone 270 can be any kind of wireless or wired connection, and may be digital or analog, infrared (IR) or transmitted radio frequencies (RF) without, departing from the scope of the invention. Jannard '526 at 8:45–47 40. Alternatively, the audio device 10A can include a wireless transceiver for receiving digital signals from another device. Jannard '526 at 20:57–64 The interface device 110 includes a power source 112, a transceiver 114, an interface 116, and an antenna 118. The power source 112 can be in the form of disposable or rechargeable batteries. Optionally, the power source 112 can be in the form of solar panels and a power regulator. The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. Jannard '526 at 21:35–39 The interface 116 can be configured to receive signals from the transceiver 114 that are in the form of digital or analog audio signals. The interface 116 can then send the audio signals to the speakers 14C, 16C through speaker lines 120, 122, respectively, discussed in greater detail below. Jannard '526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. Jannard '526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player

Claims	Prior Art
	such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann or Jannard to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including storing a hearing profile/characteristics for the user based on the results of the hearing test, i.e., such that the hearing profile/characteristics stores the appropriate hearing parameters for the user on the ear–level device itself, and using the hearing profile/characteristics to compensate for the user's hearing loss, as described above in the citations. A POSITA would have found it obvious to configure and use the hearing aid with an external device, such a mobile communication device, to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of signals received by the digital hearing aid itself. To the extent the specific details of features/functionality of similar digital hearing aids are not expressly disclosed, a POSITA thus would have been motivated to look to other references to supply these missing details. A POSITA would have had a reasonable expectation of success in implementing this combination, as wireless communication between host and ear–level devices and digital signal processing were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid.

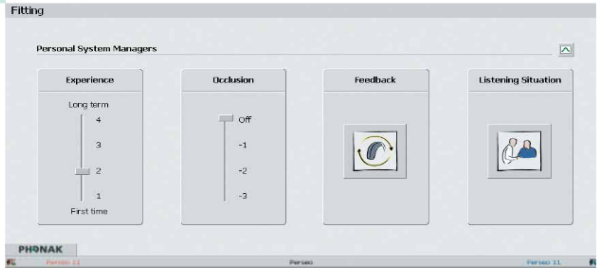
Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1c] electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to (i) store hearing characteristics associated with the user;	DHA systems, alone or in view of Anderson, Enzmann or Jannard, systems expressly or inherently disclose "electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to (i) store hearing characteristics associated with the user." Phonak – Perseo Brochure at Page 5 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 7  Phonak – Perseo Brochure at p. 17 With the development of Perseo, PFG has become significantly easier to use, providing direct access to the benefits of high-tech hearing systems. PFG for Perseo pre-sets the instrument according to each wearer's hearing ability and needs. The experience gained over a decade of PFG fittings and extensive field trials is integrated automatically into initial settings. This ensures maximum spontaneous acceptance. The wearer immediately feels confident that Perseo is the right choice. This confidence continues even if additional adjustments are necessary, thanks to the flexibility of the three levels of professional control – Easy with Personal System Managers, Standard with clustered controls, and Detailed with access to each of the 20 channels.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 18

The art of the personal touch – Personal System Managers

PFG 8.1 provides a set of **Personal System Managers** to take care of the technical details behind common fitting issues, leaving the fitter free to take care of the wearer. This approach ensures that fitting is uncomplicated. It is always clear which controls should be used and how to use them.



The **Experience Manager** adjusts tonality and loudness to match each client's experience with hearing instruments. An effective and convenient control to set and adjust the client's hearing system as they become accustomed to their new world of sound.

The **Occlusion Manager** eliminates the feeling of blocked ears experienced by some wearers.

The dynamic **Feedback Manager** automatically identifies any bands at risk from feedback. Processing strategies are modified in these bands to prevent feedback while also increasing audibility for soft sounds. The result is easier listening without feedback.

The **Listening Situation Manager** allows the fitter to program *PersonalLogic* according to the wearer's preference for when to trigger a change of base programs, and the desired degree of sound contrast between the base-programs. Additional personalization is available, if required, by directly programming either of the base programs.

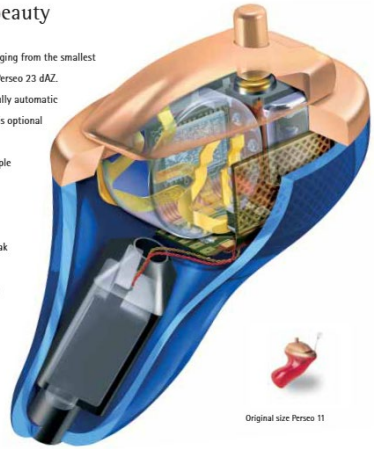
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.



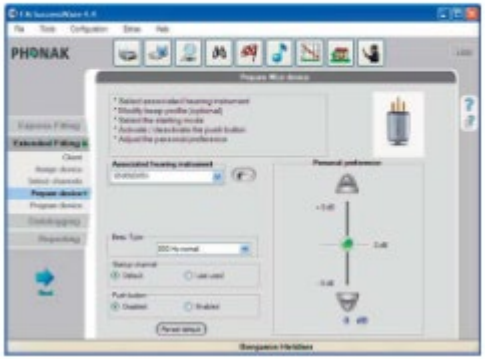
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Phonak – Online Interview re Perseo (April 7, 2003) at Page 2

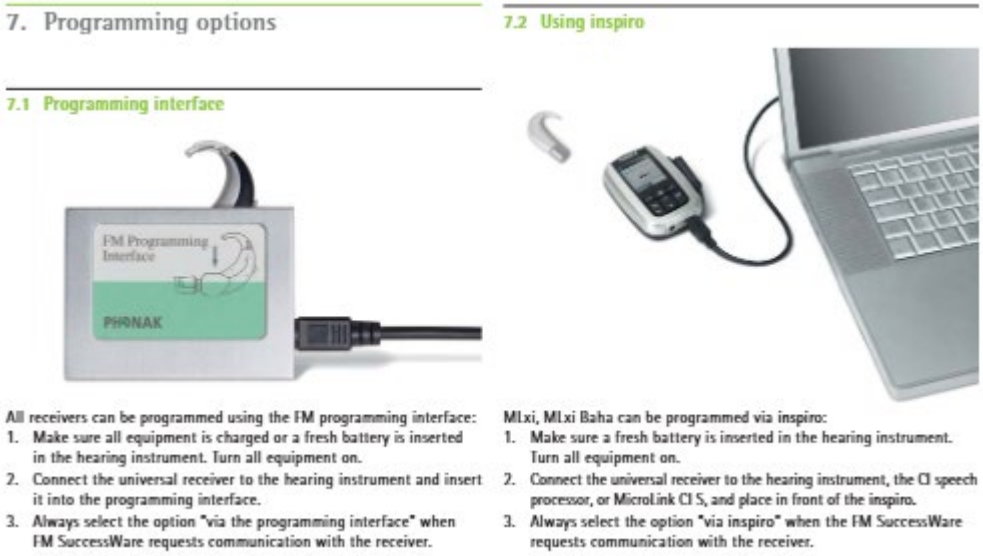
Dave Fabry/Phonak: First of all, Perseo serves as the flagship hearing system in Phonak's hearing aid family, combining leading edge, innovative hardware with efficient, flexible, and user-friendly software in one package.

We call it the art and science of hearing aid fitting, in that it enables both experienced and inexperienced clinicians to efficiently optimize hearing aid parameters for each and every patient at the initial fitting session.

Furthermore, using a series of Personal System Managers, they enable the clinician to easily accommodate the patient's changing needs over time. Hearing solutions should be individual, and Perseo offers the ultimate in personalization to achieve that goal.

Claims	Prior Art
	Phonak – Online Interview re Perseo (April 7, 2003) at Page 5 In a nutshell, Perseo is designed for new and experienced users who are looking for a hearing instrument with advanced adaptive features, enhanced sound quality, easy personalization of fitting and choice of user operation (namely, automatic, manual program control with Tactronic switch, or via remote control). The choice of one product over another is dependent on the patient's hearing loss, listening needs, and prioritization of size, cost, and available performance features. Phonak provides an entire family of digital options from Perseo, Claro, Aero, and our new digital value line that will be introduced this summer. In addition, Supero is the only digital hearing instrument designed specifically for super-power applications. Phonak – Online Interview re Perseo (April 7, 2003) at Page 7 Personal*Logic* may be personalized, via the Listening Situation Manager, to meet individual patient preferences for listening in noise. The clinician may select whether Personal*Logic* adapts to noise on the basis of audibility or comfort. Audibility changes to Noisy situation processing only when speech in noise is detected, but not with speech alone or noise alone. Comfort adapts WHENEVER noise is present that exceeds a certain dB level. Are you with me? Phonak – Fitting Guide for MLxi and MLxi Baha at Page 4 4.2 Advanced fitting of MLxi using FM SuccessWare The extended mode of FM SuccessWare enables you to fine-tune some parameters of the MLxi according to the needs of your customers. 

Claims	Prior Art
	Phonak – Fitting Guide for MLxi and MLxi Baha at Page 8



7. Programming options

7.1 Programming interface

All receivers can be programmed using the FM programming interface:

1. Make sure all equipment is charged or a fresh battery is inserted in the hearing instrument. Turn all equipment on.
2. Connect the universal receiver to the hearing instrument and insert it into the programming interface.
3. Always select the option "via the programming interface" when FM SuccessWare requests communication with the receiver.

7.2 Using inspiro

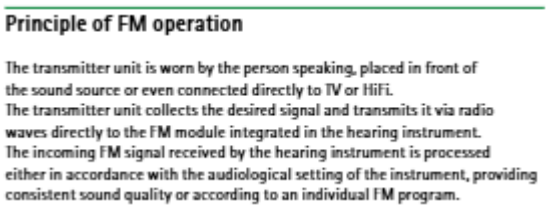
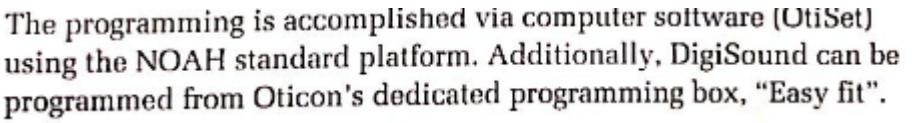
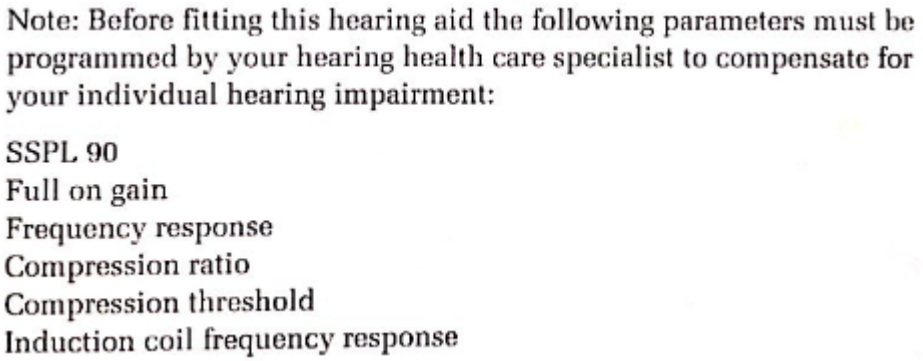
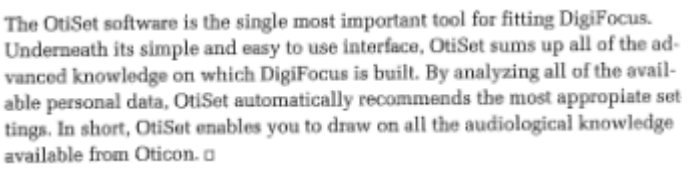
MLxi, MLxi Baha can be programmed via inspiro:

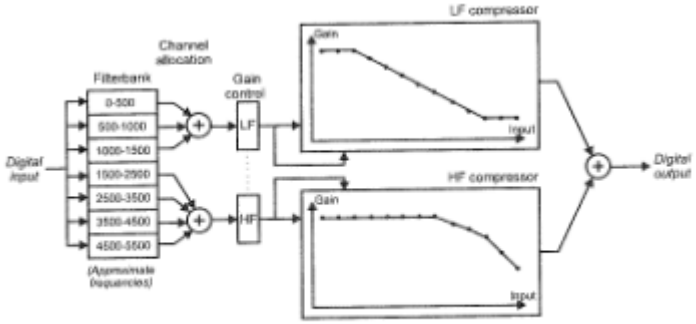
1. Make sure a fresh battery is inserted in the hearing instrument. Turn all equipment on.
2. Connect the universal receiver to the hearing instrument, the CI speech processor, or MicroLink CI S, and place in front of the inspiro.
3. Always select the option "via inspiro" when the FM SuccessWare requests communication with the receiver.

Phonak – Media Release re Product Offerings in F2003/2004 at Page 2

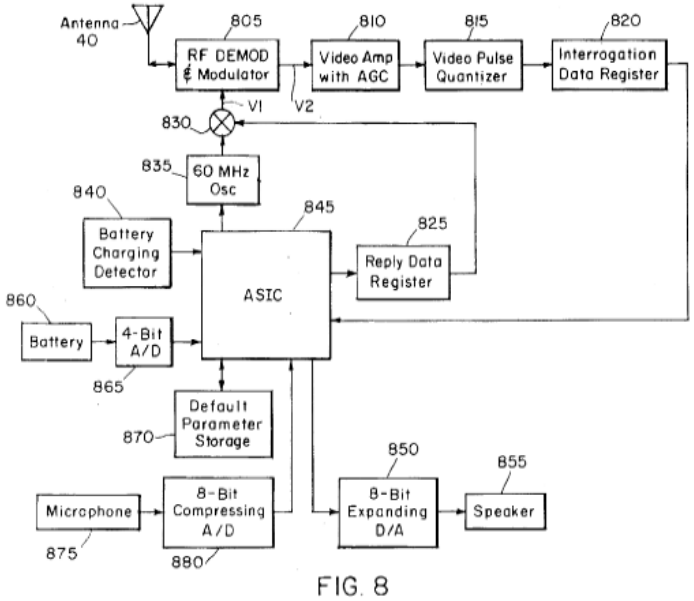
The key breakthrough technology in Perseo is Personal*Logic*, the first ever automatic control system that is fitter programmable according to the tastes and preferences of each individual wearer. Automatic performance has now a new level of intelligence beyond merely adapting to acoustic changes in the environment. Perseo is the first hearing system with customized automatic operation, as opposed to operation based on factory settings. It allows hearing professionals to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 402 The Phonak fitting software (Version 8.3) was used to program the hearing aids. First, hearing thresholds were entered into the NOAH software. The manufacturer's prescribed "First-Fit" was calculated from these hearing thresholds. The default settings in each of the software "personal system" menus (provides adjustments to the "First-Fit" for prior experience; occlusion effect and feedback) were utilized. If the patient reported occlusion, then this feature was activated. If feedback was noticed, the feedback manager was accessed. All hearing Phonak – Collections 2002 Product List at Page 33 PFG – new perspectives in fitting success   QuickGuide The versatile, easy-to-use online tour through the PFG fitting process with access to all optional fitting tools. Instrument selection wizard This online assistant recommends the product that best matches the customer's needs. Initial SoundCheck This is an ideal tool for fast and simple fine tuning of sound quality. Three controls provide convenient gain and balance settings in the high- and low-frequency ranges. Automatic fine tuning Based on the customer's hearing perception, the software automatically computes the most suitable modifications. Manual fine tuning A high degree of fitting precision can be attained with this broad spectrum of parameters in an optimised display format. Phonak Sound The degree of fitting success can be dramatically improved by using these real-life sound clips. All sound examples can be saved on a computer as MP3 files; they are also available on a sound CD. Proven PFG benefits • One program for all Phonak hearing computers • NOAH compatible or standalone version • Detailed help system with in-depth scientific and technical information • PFG training demo

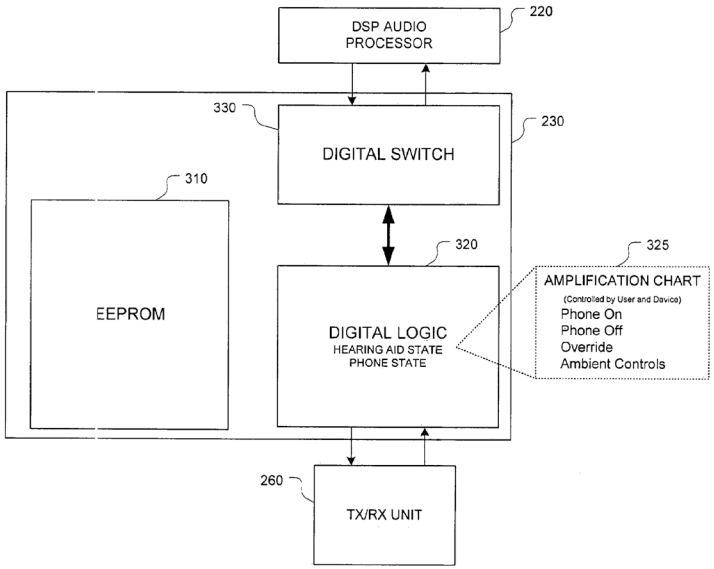
Claims	Prior Art
	Phonak – Claro 211 Product Information at Page 2  **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program. Oticon – DigiFocus Compact User Instructional Brochure at p. 2  The programming is accomplished via computer software (OtiSet) using the NOAH standard platform. Additionally, DigiSound can be programmed from Oticon's dedicated programming box, "Easy fit". Oticon – DigiFocus Compact User Instructional Brochure at p. 2  Note: Before fitting this hearing aid the following parameters must be programmed by your hearing health care specialist to compensate for your individual hearing impairment: SSPL 90 Full on gain Frequency response Compression ratio Compression threshold Induction coil frequency response Oticon – DigiFocus White Paper (1996) at Page 1  The OtiSet software is the single most important tool for fitting DigiFocus. Underneath its simple and easy to use interface, OtiSet sums up all of the advanced knowledge on which DigiFocus is built. By analyzing all of the available personal data, OtiSet automatically recommends the most appropiate settings. In short, OtiSet enables you to draw on all the audiological knowledge available from Oticon. □

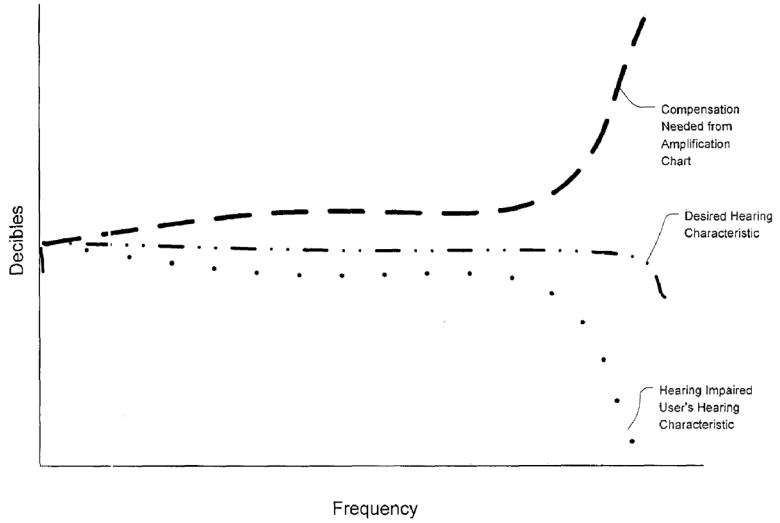
Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Fig. 1  *Fig.1. DigiFocus - block diagram of the digital audio processor, DAP.* Oticon – DigiFocus White Paper (1996) at Page 9 With DigiFocus you have an extremely flexible hearing instrument. By using the right tools you can define a personal, social and audiological profile for each individual patient. This enables you to utilize the capacity of DigiFocus to the fullest extent. The Human Link tools for DigiFocus include: Oticon – DigiFocus White Paper (1996) at Page 9 • The DigiFocus diary enables you to determine quickly when a hearing ir strument needs adjustment • The fine-tuning recommendations help you to adjust the settings • OtiSet, a PC based software program, and the programming belt enable you to fit and fine tune DigiFocus while the client is wearing the instrument. This takes place without your patient being wired to the PC.

Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 1-0 OtiSet version 3.5 is a programming and fitting software module that complies with the hearing industry's standard NOAH platform. It runs on any PC in a Windows 3.11 environment. Together with Oticon's other Human Link tools, it provides everything you need to fit DigiFocus. OtiSet has been designed to make fitting easy while giving you relevant and context-sensitive audiological information. Underneath its simple and easy to use interface, OtiSet considers all the advanced knowledge on which DigiFocus is built. It relies on top audiologi-cal research, summarizing many years of work and field testing performed at Oticon's Eriksholm Research Centre in cooperation with hearing care professionals and audiological researchers worldwide. The new version of OtiSet fully supports DigiFocus, helping you to fit the exact needs of your patients. While using OtiSet, you are able to compare your patient's audiogram with the fitting range of all Oticon's hearing instruments. Based on the choice of instrument and available audiological data, OtiSet automatically recommends the initial control settings. During fitting, OtiSet gives you instant access to graphical representations that show how various speech and sound spectra perform as the controls are adjusted on screen. Oticon – DigiFocus White Paper (1996) at Page 10 *The OtiSet software is the single most important tool for fitting DigiFocus. Underneath its simple and easy to use interface, OtiSet sums up all of the advanced knowledge on which DigiFocus is built. By analyzing all of the available personal data, OtiSet automatically recommends the best suitable settings. In short, OtiSet enables you to draw on all the audiological knowledge available from Oticon.* Oticon – DigiFocus White Paper (1996) at Page 11 Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time.

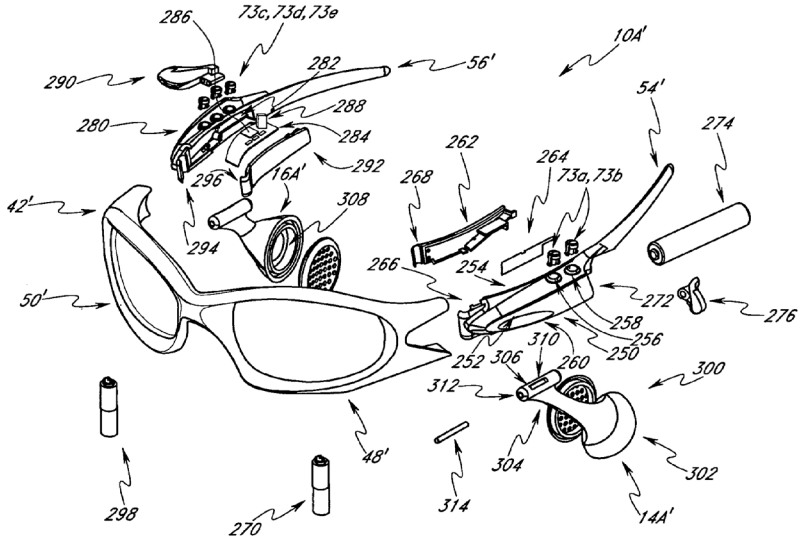
Claims	Prior Art
	Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 DigiFocus operates on two tiny new computer chips that have the processing power of a normal desktop computer. Traditional hearing instruments have only up to 10 adjustable parameters, but Oticon representatives say that their processor will give a maximum of 100 software-controlled adjustment parameters. And the device can be programmed by an audiologist while the hearing-impaired person is wearing it. Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Anderson '783 at FIG. 8  FIG. 8 Anderson '783 at 3:40–41 FIG. 8 is a block diagram showing details of an earpiece for a full-featured preferred embodiment of the

Claims	Prior Art
	invention. Anderson ’783 at 12:38–46 Note that all earpiece functions (e.g., power control for the 60 MHz oscillator 835) are controlled by a custom application-specific integrated circuit (ASIC) 845. The ASIC 845 may be a low-power low-voltage device similar, e.g., to the ECI Semiconductor P576 custom programmable linear/digital gate array. Many of the functions that are shown as separate blocks for clarity in FIG. 8, e.g., reply data register 825, may be implemented within the ASIC 845. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when

Claims	Prior Art
	the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Enzmann ’730 at FIG. 4  FIG. 4

Claims	Prior Art
	Enzmann ’730 at FIG. 5  FIG. 5 Enzmann ’730 at 5:41–44 FIG. 4 provides an expanded view of the source control module 230 according to this invention. The source control module 230 preferably includes at least a digital switch 330, digital logic 320 and a memory device 310. Enzmann ’730 at 5:59–67 The digital switch 330 operates as a double pole, two-position switch. The digital switch 330 is controlled by an input from the digital logic 320. Upon receipt of a proper input from the digital logic 320, the activation state, and thus the position, of the digital switch 330 is changed. The activation state of the digital switch 330 is either opened (OFF) or closed (ON), the latter of which permits input signals to be sent to/from the digital logic 320 to the DSP Audio processor 220. Enzmann ’730 at 6:1–11 The digital logic 320 contains a series of devices consisting of multiple identical and separately identifiable structures built of some mix of combinational logic and sequential elements the sequential elements being commonly referred to as registers, latches, flip-flops, or shift register latches. These

Claims	Prior Art
	devices individually or in combination implement a series of activation states and wave patterns that determine the activation state of the aided ear bud and shapes of the frequency wave output from the digital logic 320. The functionality of the digital logic 320 is depicted in FIG. 4 as the amplification chart 325. Enzmann ’730 at 6:12–35 The amplification chart 325 contains a series of pre-stored dynamic frequency maps based on the user's hearing characteristics, the speaker characteristics and the ambient noise near the aided ear bud 200. The digital logic 320 will receive the transmitted signal from the TX/RX unit 260 and process those signals to an optimal form based on the user's hearing, speaker and ambient noise characteristics. The arrangement of the logic devices within the digital logic 320 will serve to produce each of the characteristics (wave shape) of the amplitude chart 325. The user's hearing characteristics remain static unless the aided ear bud 200 is reprogrammed to compensate for additional hearing loss or increase and different users. For example, as shown in FIG. 5, a mapping of a user's hearing characteristic (denoted by the “••” line) illustrates that the user lacks significant hearing capability in higher frequencies. The desired hearing characteristic is denoted by the “-••” line. This desired hearing characteristic depicts or represents how a non-hearing impaired person's hearing characteristics may be. As known to those skilled in the art, close representation of this hearing characteristic is achievable by adding a complementary characteristic to the user's hearing characteristic to produce a result that is very close to the desired characteristic. Each of the characteristics is corrected using the process shown in FIG. 5. Enzmann ’730 at 6:44–48 It should be appreciated that the characteristics represented in FIG. 5 are for illustrative purposes only. Each of the characteristics in FIG. 5 are device and user specific. Each of the characteristics can be set and calculated by information stored in the EEPROM 310.

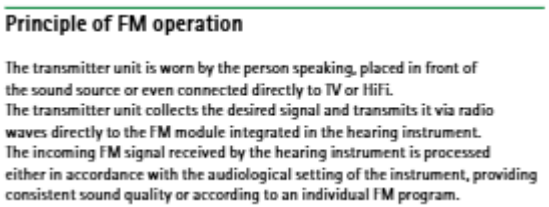
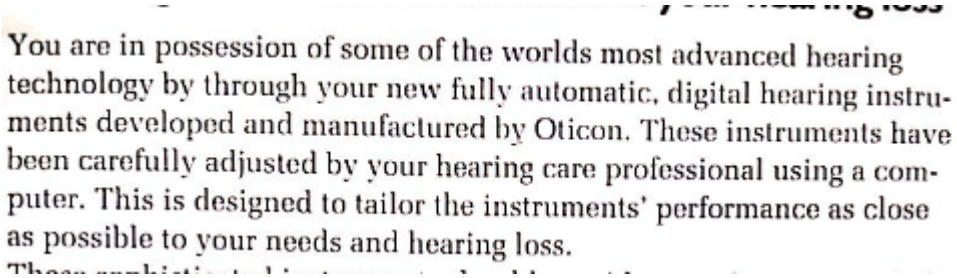
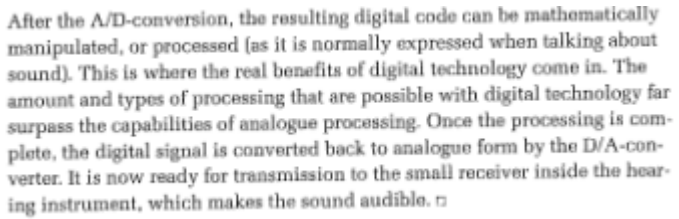
Claims	Prior Art
	Jannard ’526 at FIG. 3J  *FIG. 3J* Jannard ’526 at 3:56–58 FIG. 3J is a front, top, and left side perspective and exploded view of the wearable audio device illustrated in FIG. 3D. Jannard ’526 at 12:62–13:6 A cover member 262 cooperates with the housing 250 to define the closed internal cavity 252. In the illustrated embodiment, the internal cavity 252 includes at least one compartment configured to receive an electronic circuit board 264 which includes at least one switch for each of the buttons 73a, 73b. In an exemplary but non-limiting embodiment, the board 264 can include two switches, one for each of the buttons 73a, 73b, which are configured to control a volume output from the speakers 14A′, 16A′. The cover 262 can be attached to the ear stems 54A′ with any type of fastener, such as, for example, but without limitation, screws, rivets, bolts, adhesive, and the like. Jannard ’526 at 13:50–55 The internal cavity 282, in the illustrated embodiment, is configured to receive a printed circuit board 284. In the illustrated embodiment, the printed circuit board 284 includes one switch for each of the buttons

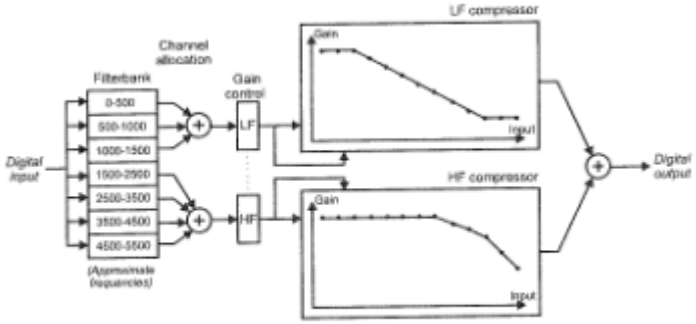
Claims	Prior Art
	73c, 73d, and 73e. Additionally, the printed circuit board 284 includes an audio file storage and playback device 286. Jannard '526 at 14:6–14 Advantageously, the printed circuit board 284 also includes or is in electrical communication with a data transfer port 388. In the illustrated embodiment, the housing 280 includes an aperture (not shown) disposed in a position similar to the position of the aperture 272 on the housing 250. In the housing 280, however, the aperture is aligned with the data transfer port 288. Thus, when the printed circuit board 284 is received in the internal cavity 282, the data transfer port 288 is aligned with the aperture. Jannard '526 at 15:60–16:2 Thus, with reference to FIGS. 3J and 3M, the ribbon connector 318 can extend from the housing 280, into the recesses 294, 296, through an aperture (not shown) in the hinge pin 298 to the upper opening within the hinge pin 298, then through the channel 316 (FIG. 3E), to an upper opening of the hinge pin 270, out through an aperture (not shown) through a side of a hinge pin 270, through the recesses 266, 268 of the housing 250, and then to the speaker 14A′, printed circuit board 264, and the power storage device 274. Jannard '526 at 25:12–21 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the audio device 10C is advantageously adapted to support any of a variety of portable electronic circuitry or devices which have previously been difficult to incorporate into conventional headsets due to bulk, weight or other considerations. For example, but without limitation, the electronics are digital or other storage devices and retrieval circuitry such as for retrieving music or other information from MP3 format memory or other memory devices. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann or Jannard to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including storing a hearing profile/characteristic for the user based on the results of the hearing test, as described above in the citations. A POSITA would have found it obvious to configure and use the hearing aid to process received wirelessly from with an external device, such as a mobile communication device, to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of the digital hearing aid. To the extent the specific details of features/functionality of storing a hearing characteristic is not expressly disclosed, a POSITA thus

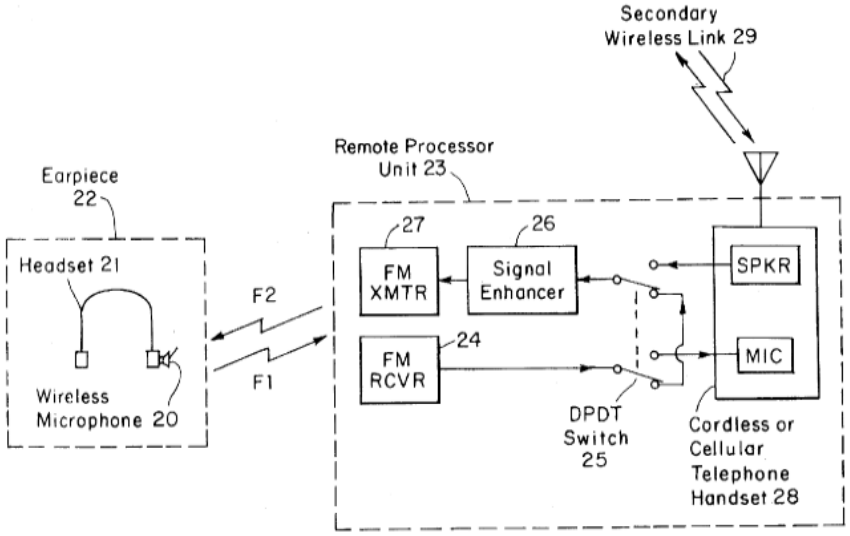
Claims	Prior Art
	would have been motivated to look to other references to supply these missing details. A POSITA would have had a reasonable expectation of success in implementing this combination as digital signal processing of audio signals to compensate for hearing loss were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1d] (ii) customize the received audio output signals to produce customized audio output signals for the user based on at least a portion of the stored hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear; and	DHA systems, alone or in combination with Anderson and Enzmann, expressly or inherently disclose "(ii) customize the received audio output signals to produce customized audio output signals for the user based on at least a portion of the stored hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear." Phonak – Perseo Brochure at Page 11 ▪ With **dual–base automatic** the fitter has direct access to the two separate base-programs. Performance can be matched to each user's needs in quiet situations without disrupting performance in noisy situations and vice versa. The result is less fine-tuning effort than with single-base instruments. With **single–base automatic** time is wasted "juggling" a single set of fitting parameters while attempting to balance performance in multiple listening situations. ▪ To further simplify fitting, Personal*Logic* is pre-set according to typical needs and is easily re-programmed to individual preferences using the **Listening Situation Manager (LSM)**. Ensuring that an automatic hearing system responds correctly in noisy situations where there is no speech is an important consideration for safety and to eliminate feelings of isolation. The **LSM** provides this functionality with a single mouse click. Matching the system's sensitivity to noise with that of the client's, and controlling the auditory contrast between quiet and noisy settings is equally easy.

Claims	Prior Art
	Phonak – Online Interview re Perseo (April 7, 2003) at Page 7 PersonalLogic may be personalized, via the Listening Situation Manager, to meet individual patient preferences for listening in noise. The clinician may select whether PersonalLogic adapts to noise on the basis of audibility or comfort. Audibility changes to Noisy situation processing only when speech in noise is detected, but not with speech alone or noise alone. Comfort adapts WHENEVER noise is present that exceeds a certain dB level. Are you with me? Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink, based on Phonak breakthrough digital FM synthesizer technology, now enables people with a hearing impairment and those with normal hearing to use the same communication tools with equal effectiveness. SmartLink is a communication gateway to all situations where understanding perfectly, all of the time, is key. The unique innovation, SmartLink, meets the needs of people with hearing loss, giving them access to effective communication in difficult situations and to the many advantages off modern communication tools, via one device. Phonak – Media Release re Product Offerings in F2003/2004 at Page 2 The key breakthrough technology in Perseo is PersonalLogic, the first ever automatic control system that is fitter programmable according to the tastes and preferences of each individual wearer. Automatic performance has now a new level of intelligence beyond merely adapting to acoustic changes in the environment. Perseo is the first hearing system with customized automatic operation, as opposed to operation based on factory settings. It allows hearing professionals to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes.

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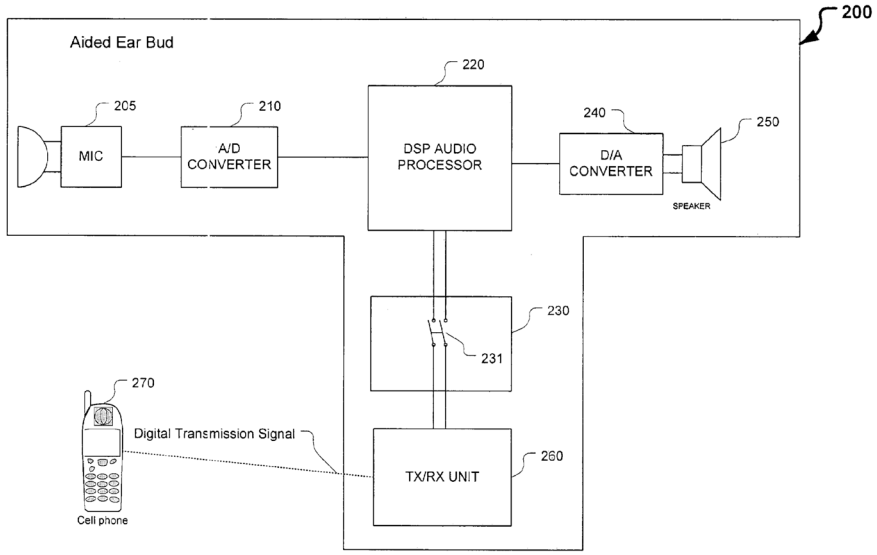
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Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Fig. 1  *Fig.1. DigiFocus - block diagram of the digital audio processor, DAP.* Oticon – DigiFocus White Paper (1996) at Page 4 The input signal from the microphone or the telecoil is digitized and fed to a filterbank, which divides the signal into seven band-pass filters with centre frequencies at: 250, 700, 1200, 2000, 3000, 4000 and 5000 Hz. The filterbank covers the frequency range from 125 to about 6000 Hz. The output from each band-pass filter is equipped with an attenuator by which each filtered signal can be weighted. This allows for a very precise fitting in accordance with the audiometric profile of the hearing impaired.

Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2 Anderson ’783 at 3:19–21 FIG. 2 is a block diagram illustrating a limited-feature preferred embodiment of the invention that uses commercial off-the-shelf system elements. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2.

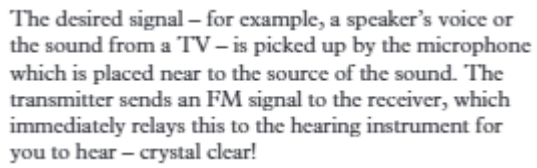
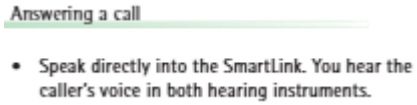
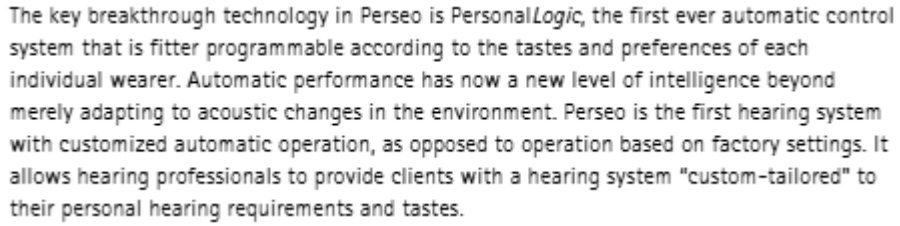
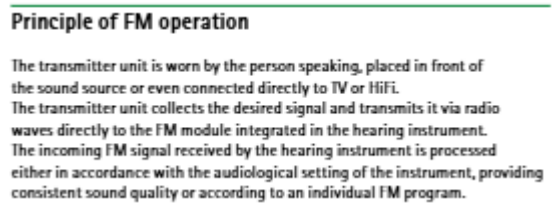
Claims	Prior Art
	Anderson ’783 at 21:40–46 If no interrogation is received by the earpiece 10 from the RPU 16 for a certain period of time (e.g., 0.1 second), as determined for example by a retriggerable monostable multivibrator circuit (also known in the art as a "one-shot") implemented in the earpiece ASIC 845, the ASIC initiates simulation of a stand-alone CIC hearing aid earpiece. Anderson ’783 at 21:46–50 Stand-alone operation is provided when the ASIC 845 causes signals from the microphone 875 via the 8-bit compressing A/D 880 to be routed to the speaker 855 via the 8-bit expanding D/A 850. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC

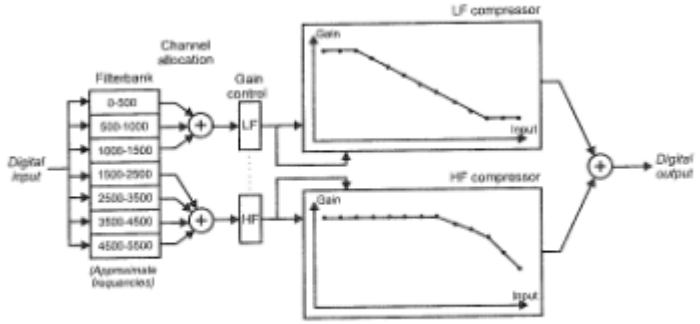
Claims	Prior Art
	default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 27:4–24 Hearing test: The present invention allows hearing tests to be performed via the earpiece normally worn by the user, without the need for additional expensive test equipment. All signal generation capabilities necessary to perform hearing tests are available in the hearing aid system comprising an earpiece 10 and RPU 16. Specifically, the RPU DSP 948 can be used to generate tones of varying frequencies and amplitudes, as well as other signals known to be useful for hearing test purposes (e.g., nonsense syllables) that are converted to acoustic waves by the earpiece. Such tones are used to perform a hearing test that determines appropriate gain vs. frequency parameters for a program stored in the RPU DSP 948 that performs signal enhancement to compensate for the user's hearing loss. Note that the hearing test program that controls the RPU DSP 948 during the hearing test can be temporarily stored in the RPU DSP 948 for the duration of the test, then deleted upon completion of the test to allow re-use of RPU DSP 948 memory resources during normal operation. The hearing test program may be loaded into the RPU DSP 948 through the secondary wireless link 944 or a wired peripheral link 950.

Claims	Prior Art
	Enzmann ’730 at FIG. 2  FIG. 2 Enzmann ’730 at 3:20–34 The conventional hearing aid comprises a microphone 105, an A/D converter 110, a digital signal audio processor (DSP) 120, a D/A converter 130 and a speaker 140. In operation, sounds, represented by an analog signal and/or ambient noise emitted from an outside source, is captured by the microphone 105 and converted to a digital signal by the A/D converter 110. The DSP 120 manipulates the converted signal in accordance with well-known techniques and sends a processed signal to the D/A converter 130. In the D/A converter 130, the processed signal is converted to an analog signal and output to the speaker 140. The speaker 140 is placed in or around the hearing impaired user's ear. Thus, the captured sounds have been captured, processed, and amplified and output to the user. Enzmann ’730 at 3:35–40 The manipulating processes of the DSP 120 serves both to amplify the converted signal from the A/D converter 110 and correct any degraded signal areas so that an improved signal is sent to the speaker 140. The manipulation processes may also adjust the incoming signal based on a preprogrammed frequency mapping of the user's hearing characteristics.

Claims	Prior Art
	Enzmann ’730 at 6:12–35 The amplification chart 325 contains a series of pre-stored dynamic frequency maps based on the user's hearing characteristics, the speaker characteristics and the ambient noise near the aided ear bud 200. The digital logic 320 will receive the transmitted signal from the TX/RX unit 260 and process those signals to an optimal form based on the user's hearing, speaker and ambient noise characteristics. The arrangement of the logic devices within the digital logic 320 will serve to produce each of the characteristics (wave shape) of the amplitude chart 325. The user's hearing characteristics remain static unless the aided ear bud 200 is reprogrammed to compensate for additional hearing loss or increase and different users. For example, as shown in FIG. 5, a mapping of a user's hearing characteristic (denoted by the “••” line) illustrates that the user lacks significant hearing capability in higher frequencies. The desired hearing characteristic is denoted by the “-••” line. This desired hearing characteristic depicts or represents how a non-hearing impaired person's hearing characteristics may be. As known to those skilled in the art, close representation of this hearing characteristic is achievable by adding a complementary characteristic to the user's hearing characteristic to produce a result that is very close to the desired characteristic. Each of the characteristics is corrected using the process shown in FIG. 5. Enzmann ’730 at 8:3–7 At step S2000, the digital signal is sent to the DSP to be processed. The process then continues to step S2100, where the corrected signal is output to the speaker of the ear bud. The speaker of the ear bud is typically positioned in or in close proximity to the user's ear. Enzmann ’730 at 8:16–23 At step S600, the aided ear bud retrieves the transmitted signal from the TX/RX unit and then proceeds to step S700. In step S700, the process fixes the transmitted signal according to signal stored in the amplification chart. Once the signal has been fixed and verified, the process proceeds to step S800, where this fixed signal is forwarded across the digital switch and fixed by the DSP. The process then continues to step S900. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann or Jannard to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including customizing audio output signals for the used based on stored hearing characteristics and the audio output signals in order to compensate for the user’s hearing impairment, as described above in the citations. A POSITA would have found it obvious to configure and use

Claims	Prior Art
	the hearing aid with an external device, such a mobile communication device to perform these operations, to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of the digital hearing aid. To the extent the specific details of features/functionality of customization performed by similar digital hearing aids are not expressly disclosed, a POSITA thus would have been motivated to look to other references to supply these missing details. A POSITA would have had a reasonable expectation of success in implementing this combination as digital signal processing of audio signals to compensate for hearing loss were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1e] (iii) output the customized audio output signals to the at least one speaker,	DHA systems, alone or in view of Anderson, Enzmann or Jannard, expressly or inherently disclose "(iii) output the customized audio output signals to the at least one speaker." Phonak – Perseo Brochure at Page 13 Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer.

Claims	Prior Art
	Phonak – Microlink User Guide at Page 2  The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak - SmartLink SX User Guide at p. 39  Answering a call • Speak directly into the SmartLink. You hear the caller's voice in both hearing instruments. Phonak – Media Release re Product Offerings in F2003/2004 at Page 2  The key breakthrough technology in Perseo is Personal*Logic*, the first ever automatic control system that is fitter programmable according to the tastes and preferences of each individual wearer. Automatic performance has now a new level of intelligence beyond merely adapting to acoustic changes in the environment. Perseo is the first hearing system with customized automatic operation, as opposed to operation based on factory settings. It allows hearing professionals to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes. Phonak – Claro 211 Product Information at Page 2  **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program.

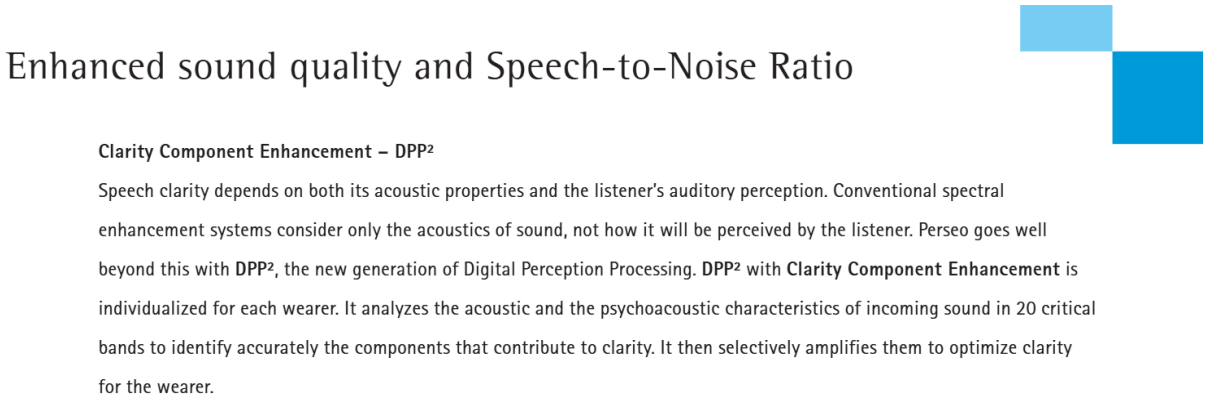
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 2 You are in possession of some of the worlds most advanced hearing technology by through your new fully automatic, digital hearing instruments developed and manufactured by Oticon. These instruments have been carefully adjusted by your hearing care professional using a computer. This is designed to tailor the instruments' performance as close as possible to your needs and hearing loss. Oticon – DigiFocus White Paper (1996) at Page 3 After the A/D-conversion, the resulting digital code can be mathematically manipulated, or processed (as it is normally expressed when talking about sound). This is where the real benefits of digital technology come in. The amount and types of processing that are possible with digital technology far surpass the capabilities of analogue processing. Once the processing is complete, the digital signal is converted back to analogue form by the D/A-converter. It is now ready for transmission to the small receiver inside the hearing instrument, which makes the sound audible. □ Oticon – DigiFocus White Paper (1996) at Fig. 1  *Fig.1. DigiFocus - block diagram of the digital audio processor, DAP.*

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Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 4 The input signal from the microphone or the telecoil is digitized and fed to a filterbank, which divides the signal into seven band-pass filters with centre frequencies at: 250, 700, 1200, 2000, 3000, 4000 and 5000 Hz. The filterbank covers the frequency range from 125 to about 6000 Hz. The output from each band-pass filter is equipped with an attenuator by which each filtered signal can be weighted. This allows for a very precise fitting in accordance with the audiometric profile of the hearing impaired. Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Anderson ’783 at 21:46–50 Stand-alone operation is provided when the ASIC 845 causes signals from the microphone 875 via the 8-bit compressing A/D 880 to be routed to the speaker 855 via the 8-bit expanding D/A 850. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Enzmann ’730 at 3:35–40 The manipulating processes of the DSP 120 serves both to amplify the converted signal from the A/D converter 110 and correct any degraded signal areas so that an improved signal is sent to the speaker 140. The manipulation processes may also adjust the incoming signal based on a preprogrammed frequency mapping of the user's hearing characteristics.

Claims	Prior Art
	Enzmann ’730 at 7:15–22 Once the digital switch is closed, signals from the TX/RX unit 260 are fixed by the digital logic 320 using the complementary characteristics, as discussed above, and passed to the DSP audio processor 220. The DSP audio processor 220 then fixes and combines a reduced representation of the input from the microphone and outputs the combined signal to the user's speaker. Enzmann ’730 at 8:3–7 At step S2000, the digital signal is sent to the DSP to be processed. The process then continues to step S2100, where the corrected signal is output to the speaker of the ear bud. The speaker of the ear bud is typically positioned in or in close proximity to the user's ear. Jannard ’526 at 8:38–42 The audio device 10A can be used as an audio output device for any type of device which provides an audio output signal. The audio device 10A can include an audio input terminal disposed anywhere on the eyeglass 40 for receiving a digital or analog audio signal. Jannard ’526 at 25:12–21 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the audio device 10C is advantageously adapted to support any of a variety of portable electronic circuitry or devices which have previously been difficult to incorporate into conventional headsets due to bulk, weight or other considerations. For example, but without limitation, the electronics are digital or other storage devices and retrieval circuitry such as for retrieving music or other information from MP3 format memory or other memory devices. Jannard ’526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204.

Claims	Prior Art
	As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann or Jannard to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including outputting customized audio output signals to the hearing aid's speaker, as described above in the citations. A POSITA would have found it obvious to configure and use the hearing aid with an external device, such as a mobile communication device to perform these operations, to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of the digital hearing aid. To the extent the specific details of features/functionality of outputting customized audio signals by similar digital hearing aids are not expressly disclosed, a POSITA thus would have been motivated to look to other references to supply these missing details. A POSITA would have had a reasonable expectation of success in implementing this combination as digital signal processing of audio signals to compensate for hearing loss were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid.

Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[1f] wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals.	DHA systems, alone or in combination with Anderson or Enzmann or Jannard, expressly or inherently disclose "wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals." Phonak – Perseo Brochure at Page 13  Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer. Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear!

Claims	Prior Art
	Phonak - SmartLink SX User Guide at p. 32  Phonak - SmartLink SX User Guide at p. 39 Answering a call • Speak directly into the SmartLink. You hear the caller's voice in both hearing instruments. Phonak - SmartLink SX User Guide at p. 58 The user benefit of FM is an improved signal-to-noise ratio, which leads to an optimized speech understanding even in very difficult listening situations.

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

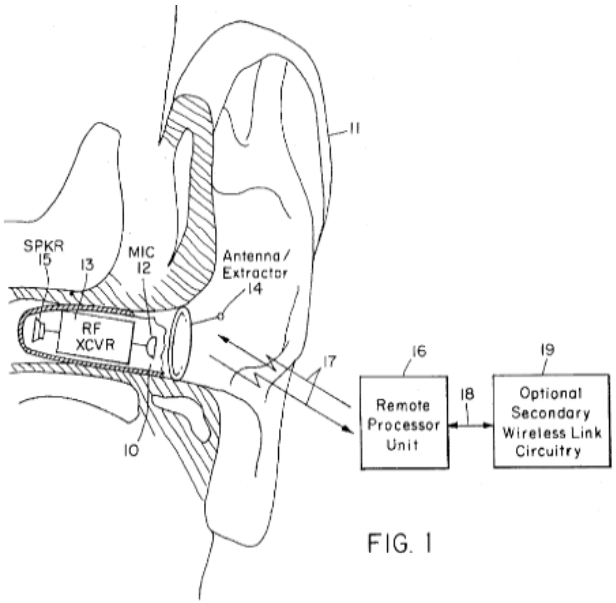
Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7  Adaptive digital AudioZoom (dAZ) Fine-scale Noise Canceler (FNC) Active and adaptive separation of speech from noise The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically. Fully automatic, fully adaptive With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR. Twenty-band noise control Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal. Precise and selective All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected. Intelligent The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved. 7 Phonak – Claro 211 Product Information at Page 2 **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program. Oticon – DigiFocus White Paper (1996) at Page 3 After the A/D-conversion, the resulting digital code can be mathematically manipulated, or processed (as it is normally expressed when talking about sound). This is where the real benefits of digital technology come in. The amount and types of processing that are possible with digital technology far surpass the capabilities of analogue processing. Once the processing is complete, the digital signal is converted back to analogue form by the D/A-converter. It is now ready for transmission to the small receiver inside the hearing instrument, which makes the sound audible. □

Claims	Prior Art
	Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Anderson '783 at FIG. 1  FIG. 1 Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to

Claims	Prior Art
	ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 4:30–33 Although the system described here is monaural, it will be clear to those skilled in the art that a binaural (stereo) system can be constructed based on the same principles as the monaural system. Anderson '783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system,

Claims	Prior Art
	e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired.

Claims	Prior Art
	Enzmann ’730 at 3:20–34 The conventional hearing aid comprises a microphone 105, an A/D converter 110, a digital signal audio processor (DSP) 120, a D/A converter 130 and a speaker 140. In operation, sounds, represented by an analog signal and/or ambient noise emitted from an outside source, is captured by the microphone 105 and converted to a digital signal by the A/D converter 110. The DSP 120 manipulates the converted signal in accordance with well-known techniques and sends a processed signal to the D/A converter 130. In the D/A converter 130, the processed signal is converted to an analog signal and output to the speaker 140. The speaker 140 is placed in or around the hearing impaired user's ear. Thus, the captured sounds have been captured, processed, and amplified and output to the user. Enzmann ’730 at 3:46–49 The embodiments of the present invention allow the user to hear the audio transmissions of the wireless phone in the speaker of the aided ear bud. This is achieved by transmitting the audio output to the aided ear bud. Enzmann ’730 at 3:55–57 In addition, the portability of the wireless phone is increased because the user's hearing device can now act as a wireless ear bud for the wireless phone. Enzmann ’730 at 4:26–29 Ambient noise and the user's voice are captures by the microphone 205 and processed and combined with signals from the TX/RX unit 260 in the DSP 220 for output through the D/A converter 246 and the speaker 250. Enzmann ’730 at 5:34–40 The signal 239, output to the speaker 250, is a combined signal based on inputs from the microphone 205 and the wireless phone 270. The combined signal is made to duplicate the experience of a non-hearing impaired person, in that the non-hearing impaired person hears a combination of his or her own voice and ambient noise when speaking into a wireless phone. Jannard ’526 at 19:48–54 The user U can carry a “body borne” source device B such as, for example, but without limitation, a cellular phone, an MP3 player, a “two-way” radio, a palmtop computer, or a laptop computer. As such,

Claims	Prior Art
	the user U can use the audio device 10C to receive and listen to audio signals from the source device B, and/or transmit audio signals to the source device B. Jannard '526 at 20:1–5 For example, but without limitation, where the source device B is a cellular phone, the user U can listen to the audio output of the cellular phone, such as the voice of a caller, through sound transducers in the audio device 10C. Jannard '526 at 21:35–39 The interface 116 can be configured to receive signals from the transceiver 114 that are in the form of digital or analog audio signals. The interface 116 can then send the audio signals to the speakers 14C, 16C through speaker lines 120, 122, respectively, discussed in greater detail below. Jannard '526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204.

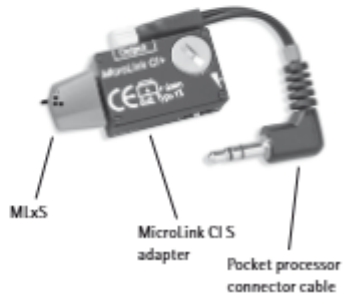
Claims	Prior Art
	The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. Jannard '526 at 28:52–64 With reference to FIG. 19, a more detailed illustration of the transceiver 114 is illustrated therein. As shown in FIG. 19, the transceiver includes a preamplifier 230, a band pass filter 232, and an amplifier 234 connected in series. The preamplifier 230 and the amplifier 234 can be of any known type, as known to those of ordinary skill in the art. The band filter 232, in the present embodiment, can be constructed as a band pass filter, allowing signals having a frequency from 48 MHz to 50 MHz, inclusive, to pass therethrough. Alternatively, the band filter 232 can be comprised of three band pass filters configured to allow frequencies centered on 48 MHz, 49 MHz, and 50 MHz, respectively, pass therethrough. Jannard '526 at 29:9–19 As the detectors 242, 244 detect 48 MHz and 50 MHz detectors, respectively, they output signals to a spread spectrum pattern detector 250. The spread spectrum pattern detector outputs a corresponding signal to a serial-to-parallel converter 252. The output of the serial-to-parallel converter 252 is output to a digital-to-analog converter 204. A "class D" audio amplifier (not shown), for example, but without limitation, can be connected to the output of the digital-to-audio converter 204 to thereby supply an audio signal to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann or Jannard to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including the hearing's aid speaker generating a portion of the customized audio output, as described above in the citations. A POSITA would have found it obvious to configure and use the hearing aid with an external device, such as a mobile communication device to perform these operations, to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of the digital hearing aid. To the extent the specific details of features/functionality of outputting customized audio signals by similar digital hearing aids are not expressly disclosed, a POSITA thus would have been motivated to look to other references to supply these missing details. A POSITA would have had a reasonable expectation of success in implementing this combination as digital signal processing of audio signals to compensate for hearing loss were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid.

Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
2. A head-worn personal electronic apparatus as recited in claim 1, wherein the received audio output signals are wirelessly received from a mobile communication device associated with the user.	DHA systems, alone or in combination with Anderson or Enzmann or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the received audio output signals are wirelessly received from a mobile communication device associated with the user." Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak – Microlink User Guide at Page 7 You may even connect the system directly to your telephone, if it has an audio output socket. Alternatively, use the optional adapter available for your HandyMic. Switching off the microphone then allows you to hear the telephone signal without any background noise. Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results.

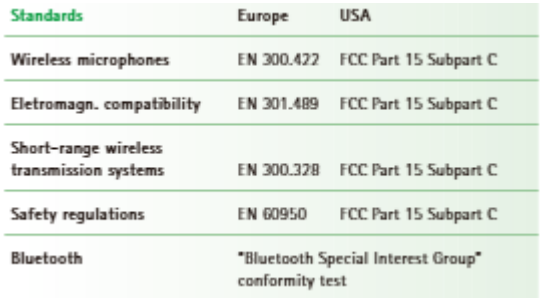
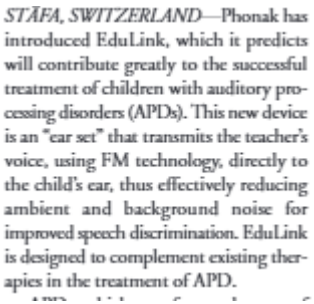
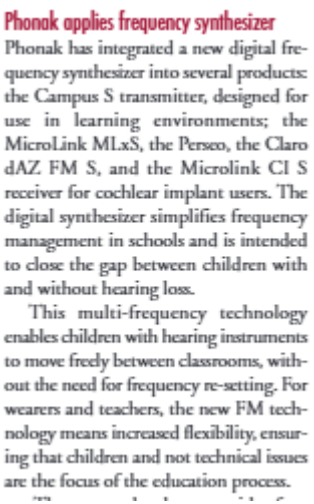
Claims	Prior Art
	Phonak – Online Interview re Perseo (April 7, 2003) at Page 9 **Liz Brassine/Phonak:** It is a digital microphone on the CIC. There are several advantages of a digital microphone over analog mics. First, increased dynamic range is provided. Additionally, with a digital mic, the analog-to-digital converter is housed within the microphone itself. This eliminates the wire between the AD-converter and the microphone. The wire is the source for ultra-high frequency interference, such as Bluetooth. Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink SX is the first communication device to incorporate • Advanced FM features • Hearing Instrument remote control • A Bluetooth link for cell phone use Phonak - SmartLink SX - FM Systems Manual at p. 4 Smart connections to all audio sources Any external audio input or an external microphone signal can be transmitted wirelessly via SmartLink to the hearing instrument, simply by attaching to the smart interface. Phonak - SmartLink SX - FM Systems Manual at p. 10 Smart links with Bluetooth With the inclusion of Bluetooth technology in SmartLink, Phonak now enables hearing instrument wearers to enjoy unrestricted use of all Bluetooth compatible cellular phones. One of the major complaints of hearing impaired people is now history – talking on the telephone is even easier for them than for people with normal hearing. Simply synchronizing the Bluetooth telephone and SmartLink creates the communication path. As long as the cell phone is within a 10-meter range of SmartLink, all incoming calls are automatically directed via SmartLink to the hearing instrument. The cell phone is only needed for dialing outgoing calls. Whenever the cell phone is switched on, the Bluetooth function remains active for incoming calls, regardless of whether SmartLink is switched on or off. This guarantees that telephone calls are not missed, even when the FM function is not required. When using an external audio input to listen to music, incoming telephone calls have priority when the cell phone is switched on. The audio signal is automatically decreased so that the user hears the ringing tone clearly.

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 10wire  0 Phonak - SmartLink SX - FM Systems Manual at p. 11 Smart digital synthesizer technology Digital frequency synthesizer technology, only available with Phonak FM communication systems, enables online frequency synchronization and scanning of the frequencies used. Once a frequency is selected on the SmartLink, it automatically sends a signal to the FM receiver that adjusts the frequency of the receiver accordingly. This automatic synchronization process occurs every time the SmartLink is switched on. In cases where more than one FM transmitter is in use, at lectures or presentations for example, the SmartLink has an exclusive scan function. This seeks out the frequency being used by the other transmitter and sets the receiver frequency accordingly.

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 14  Phonak - SmartLink SX User Guide at Page 31 • SmartLink can communicate with a Bluetooth compatible mobile phone. Please ask your mobile phone dealer if you are uncertain whether your model is Bluetooth-capable.

Claims	Prior Art
	Phonak - SmartLink SX User Guide at p. 32  Phonak MicroLink MLxS Product Information at Page 1 The MicroLink MLxS is the first miniaturized FM receiver with a digital frequency synthesizer for full multi-frequency flexibility. The receiver features fully automatic, non-contact frequency synchronization with either a WallPilot or the Campus S. The MicroLink MLxS is compatible with all Phonak FM transmitters and the WallPilot, and is recommended for use with this equipment. **MLxS with BTE instrument** **MLxS with adapter for CI pocket processor**  

Claims	Prior Art
	Phonak – Media Release re Product Offerings in F2003/2004 at Page 3 **Phonak Communications introduces the new MicroLink receivers with frequency synthesizer** Phonak Communications will introduce its new multi frequency MLx S and ML8 S receivers in the next few months. MLx S and ML8 S are the counterpart of Campus S transmitters, already introduced early 2002. Together, they comprise the new synthesized FM system generation, a system that transmits, without wires, speech to the hearing instrument in difficult listening situations. The key technology of the new system is its easy frequency adaptation. Given the extreme design constraints (low voltage, low power, miniature dimensions), the development of this technology represents an unprecedented breakthrough in wireless communication technology. Phonak SmartLink SX Product Information at p. 1 Product Information SmartLink™ SX  SmartLink SX is a compact communication device for people with hearing loss who lead busy, active lives. The SmartLink SX incorporates an FM transmitter with digital signal processing, a remote control for Phonak hearing instruments, and Bluetooth for communication with mobile phones. With its digital frequency synthesizer, the SmartLink allows a highly flexible choice of operating frequency. For maximum customer benefit, we recommend using a Phonak multi-channel FM receiver (e.g. the MLxS or ML8S) that can be automatically synchronized from the SmartLink.

Claims	Prior Art
	Phonak SmartLink SX Product Information at p. 2  Phonak and Other DHA – Manufacturers News Article at Page 1  Phonak and Other DHA – Manufacturers News Article at Page 2 

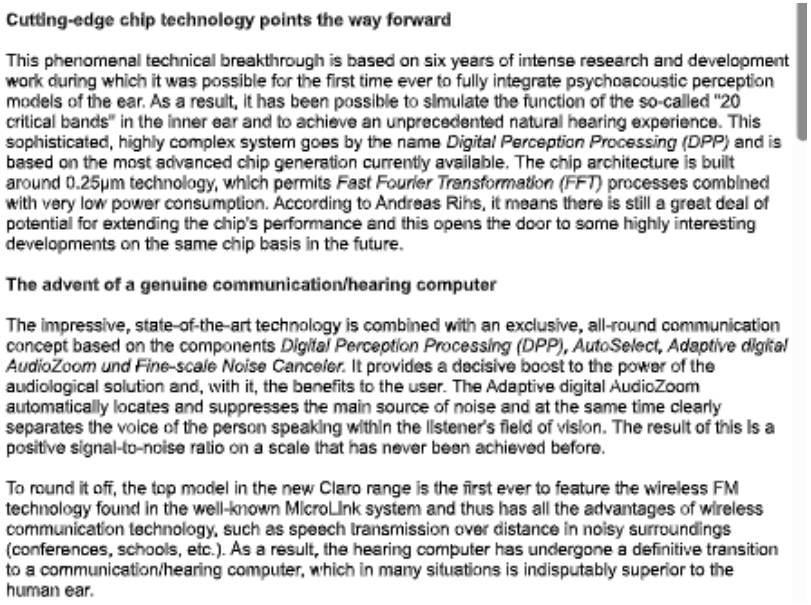
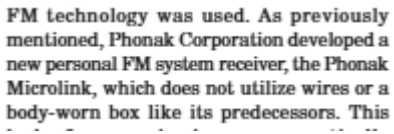
Standards	Europe	USA
Wireless microphones	EN 300.422	FCC Part 15 Subpart C
Eletromagn. compatibility	EN 301.489	FCC Part 15 Subpart C
Short-range wireless transmission systems	EN 300.328	FCC Part 15 Subpart C
Safety regulations	EN 60950	FCC Part 15 Subpart C
Bluetooth	"Bluetooth Special Interest Group" conformity test	

STÄFA, SWITZERLAND—Phonak has introduced EduLink, which it predicts will contribute greatly to the successful treatment of children with auditory processing disorders (APDs). This new device is an "ear set" that transmits the teacher's voice, using FM technology, directly to the child's ear, thus effectively reducing ambient and background noise for improved speech discrimination. EduLink is designed to complement existing therapies in the treatment of APD.

Phonak applies frequency synthesizer

Phonak has integrated a new digital frequency synthesizer into several products: the Campus S transmitter, designed for use in learning environments; the MicroLink MLxS, the Perseo, the Claro dAZ FM S, and the Microlink CI S receiver for cochlear implant users. The digital synthesizer simplifies frequency management in schools and is intended to close the gap between children with and without hearing loss.

This multi-frequency technology enables children with hearing instruments to move freely between classrooms, without the need for frequency re-setting. For wearers and teachers, the new FM technology means increased flexibility, ensuring that children and not technical issues are the focus of the education process.

Claims	Prior Art
	Phonak Claro – World Premiere at Page 1  **Cutting-edge chip technology points the way forward** This phenomenal technical breakthrough is based on six years of intense research and development work during which it was possible for the first time ever to fully integrate psychoacoustic perception models of the ear. As a result, it has been possible to simulate the function of the so-called "20 critical bands" in the inner ear and to achieve an unprecedented natural hearing experience. This sophisticated, highly complex system goes by the name *Digital Perception Processing (DPP)* and is based on the most advanced chip generation currently available. The chip architecture is built around 0.25µm technology, which permits *Fast Fourier Transformation (FFT)* processes combined with very low power consumption. According to Andreas Rihs, it means there is still a great deal of potential for extending the chip's performance and this opens the door to some highly interesting developments on the same chip basis in the future. **The advent of a genuine communication/hearing computer** The impressive, state-of-the-art technology is combined with an exclusive, all-round communication concept based on the components *Digital Perception Processing (DPP), AutoSelect, Adaptive digital AudioZoom und Fine-scale Noise Canceler.* It provides a decisive boost to the power of the audiological solution and, with it, the benefits to the user. The Adaptive digital AudioZoom automatically locates and suppresses the main source of noise and at the same time clearly separates the voice of the person speaking within the listener's field of vision. The result of this is a positive signal-to-noise ratio on a scale that has never been achieved before. To round it off, the top model in the new Claro range is the first ever to feature the wireless FM technology found in the well-known MicroLink system and thus has all the advantages of wireless communication technology, such as speech transmission over distance in noisy surroundings (conferences, schools, etc.). As a result, the hearing computer has undergone a definitive transition to a communication/hearing computer, which in many situations is indisputably superior to the human ear. Lewis – Speech perception in noise: directional microphones (2004) at Page 429  **FM technology was used. As previously mentioned, Phonak Corporation developed a new personal FM system receiver, the Phonak Microlink, which does not utilize wires or a body-worn box like its predecessors. This**

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

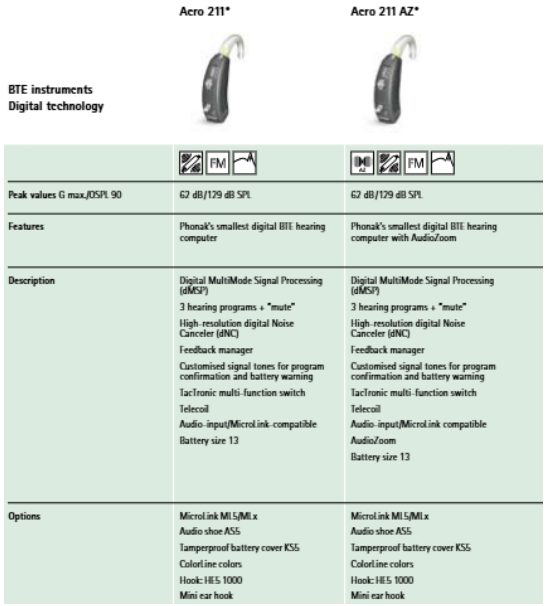
Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 9

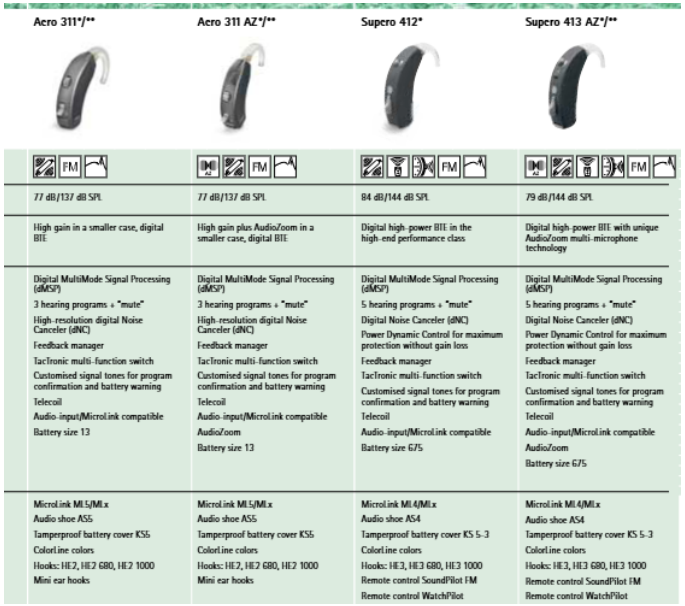
BTE instruments	Claro 111 dAZ Claro 111 dAZ FM	Claro 211 dAZ Claro 211 dAZ FM	Claro 311 dAZ Claro 311 dAZ FM
			
	FM	FM	FM
Peak values G max./OSPL 90	52 dB/126 dB SPL	62 dB/135 dB SPL	72 dB/135 dB SPL
Features	Specially designed for steep high-frequency and mild hearing losses	Digital speech transparency	Digital speech transparency and higher gain
Description	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (111 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (211 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (311 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection
Options	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)
Programming	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 16



BTE instruments Digital technology	Acro 211*	Acro 211 AZ*
	FM	FM
Peak values G max/OSPL 90	62 dB/129 dB SPL	62 dB/129 dB SPL
Features	Phonak's smallest digital BTE hearing computer	Phonak's smallest digital BTE hearing computer with AudioZoom
Description	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink-compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink compatible AudioZoom Battery size 13
Options	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 17

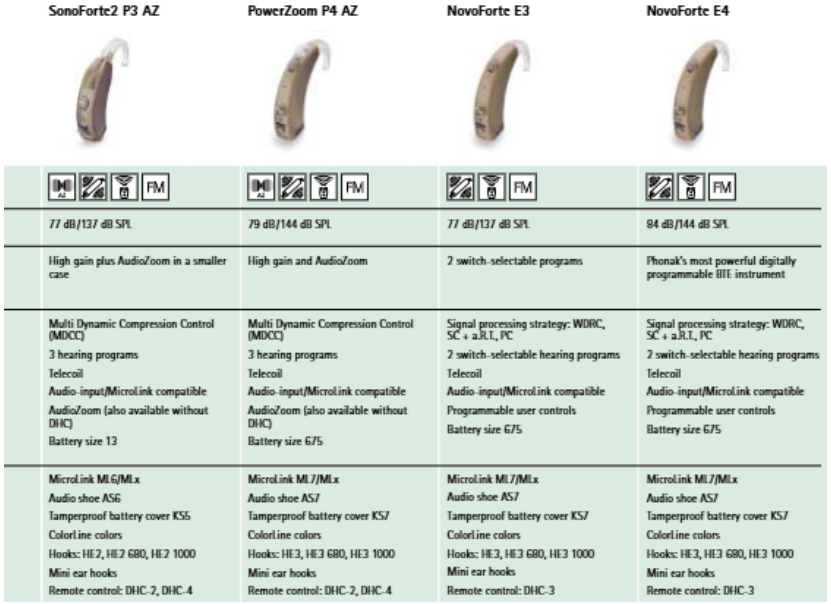


Aero 311*/**	Aero 311 AZ*/**	Supero 412*	Supero 413 AZ*/**
77 dB/137 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL	79 dB/144 dB SPL
High gain in a smaller case, digital BTE	High gain plus Audio/zoom in a smaller case, digital BTE	Digital high-power BTE in the high-end performance class	Digital high-power BTE with unique Audio/zoom multi-microphone technology
Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 13	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 675
MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 18

Digitally programmable BTE instruments	Piconet2 P2 	Piconet2 P2 AZ 
	FM	FM
Peak values G max/OSPL 90	62 dB/128 dB SPL	62 dB/128 dB SPL
Features	Phonak's smallest digitally programmable BTE hearing computer	Phonak's smallest digitally programmable BTE hearing computer with AudioZoom
Description	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13
Options	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 19

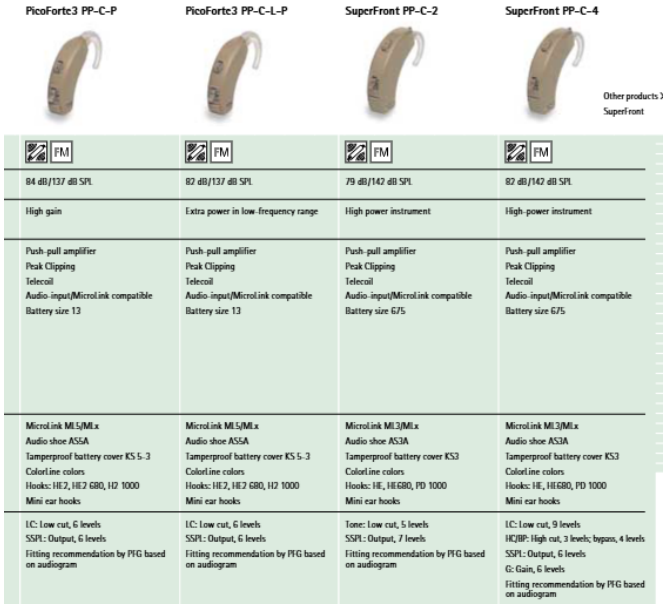


	SonoForte2 P3 AZ	PowerZoom P4 AZ	NovoForte E3	NovoForte E4
	AZ FM	AZ FM	FM	FM
	77 dB/137 dB SPL	79 dB/144 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL
	High gain plus AudioZoom in a smaller case	High gain and AudioZoom	2 switch-selectable programs	Phonak's most powerful digitally programmable BTE instrument
	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675
	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS6 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 24

BTE instruments	Supero 411*	PicoForte3 SC-D	PicoForte3 PP-SC
			
	FM	FM	FM
Peak values G max./OSPL 90	79 dB/142 dB SPL	64 dB/127 dB SPL	78 dB/133 dB SPL
Features	Digital high-power BTE	Directional microphone, switchable	High power
Description	dLimiting signal processing strategy 1 hearing program + "mute" Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13	Super Compression + a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13
Options	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, H3 1000	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks
Programming	Programmable with PC (IBM-compatible) and HI-PRO programming interface PFG Software, version 8.0 or above	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram

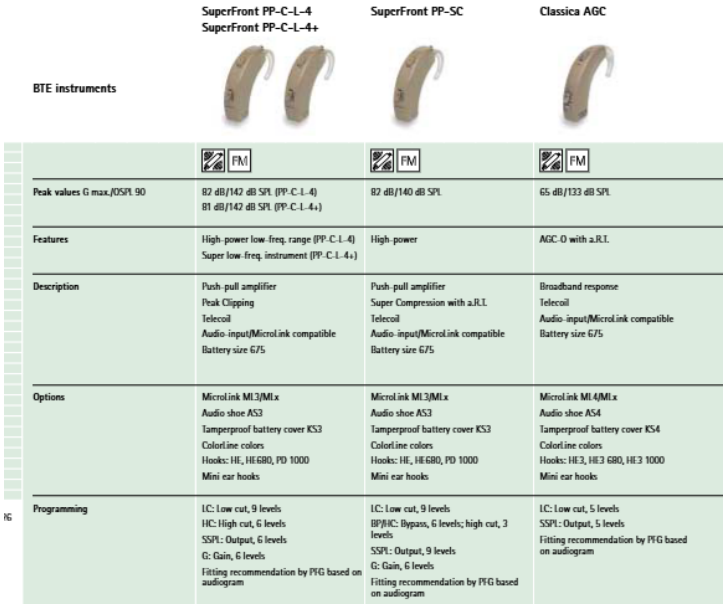
Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 25



PicoForte3 PP-C-P	PicoForte3 PP-C-L-P	SuperFront PP-C-2	SuperFront PP-C-4
			Other products) Superl·ront
FM	FM	FM	FM
84 dB/137 dB SPL	82 dB/137 dB SPL	79 dB/142 dB SPL	82 dB/142 dB SPL
High gain	Extra power in low-frequency range	High power instrument	High-power instrument
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks
LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	Tone: Low cut, 5 levels SSPL: Output, 7 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels HC/BP: High cut, 3 levels; bypass, 4 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram

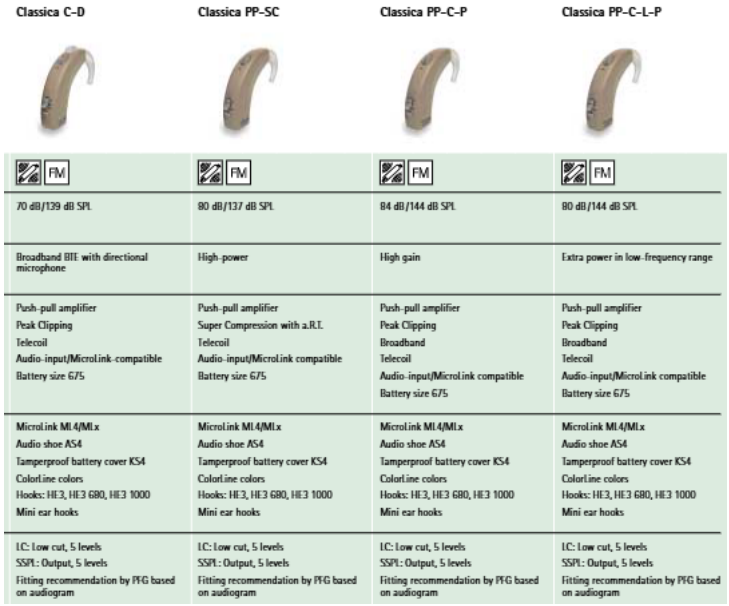
IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 135

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 26



	SuperFront PP-C-L-4 SuperFront PP-C-L-4+	SuperFront PP-SC	Classica AGC
BTE instruments			
	FM	FM	FM
Peak values G max./OSPL 90	82 dB/142 dB SPL (PP-C-L-4) 81 dB/142 dB SPL (PP-C-L-4+)	82 dB/140 dB SPL	65 dB/133 dB SPL
Features	High-power low-freq. range (PP-C-L-4) Super low-freq. instrument (PP-C-L-4+)	High-power	AGC-O with a.R.T.
Description	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Broadband response Telecoil Audio-input/MicroLink compatible Battery size 675
Options	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
Programming	LC: Low cut, 9 levels HC: High cut, 6 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels BP/HC: Bypass, 6 levels; high cut, 3 levels SSPL: Output, 9 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 27



Classica C-D	Classica PP-SC	Classica PP-C-P	Classica PP-C-L-P
FM	FM	FM	FM
70 dB/139 dB SPL	80 dB/137 dB SPL	84 dB/144 dB SPL	80 dB/144 dB SPL
Broadband BTE with directional microphone	High-power	High gain	Extra power in low-frequency range
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink-compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink ML4/ML x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/ML x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/ML x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/ML x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 29



Phonak's FM solutions:
For maximum speech intelligibility

Whenever even latest-generation hearing instruments approach their physical limits – noise, reverberation, or distance – count on Phonak's FM solutions to deliver maximum speech intelligibility.

The frequent difficulties encountered by hearing instrument users when watching TV and making phone calls can be overcome with Phonak's FM technology.

Phonak FM solutions improve the SNR by 12 to 20 dB.

The FM systems by Phonak can be used with the majority of commercially available BTE instruments and cochlear implants.

Recommended applications:

Dialog:
For example at home, on a busy street, at the shopping mall or in a restaurant.

Group discussions:
For example at a family get-together, in a very noisy restaurant, at business meetings or conferences.

Making telephone calls:
The Phonak FM system provides access to the many capabilities of modern telecommunications devices.

Watching TV and listening to music:
Phonak FM provides access to the entire spectrum of today's information and entertainment technologies.

Schools and continuing education:
For example in classes at schools and universities, on outings and excursions, in lectures or on museum tours.

Transportation:
For example in cars, trains, trams and buses.

Sports of all kinds:
For example when cycling, playing golf, skiing, jogging or hiking.

Events:
For example at the theater, in concerts, at the movies, in church services or at community meetings.

Claims	Prior Art

Phonak – Collections 2002 Product List at Page 30

MLx, the most flexible FM receiver ever made, can be used not only with Phonak BTEs but also with most third-party hearing instruments available on the market today. This is possible thanks to the standard Euro connector which simply plugs into the audio shoe of the hearing instrument.

Once again, Phonak demonstrates its commitment to innovative developments that serve hearing-impaired adults and children.

MLx compatible BTE makes:

- Argosy
- Audifon
- Beltone
- Danavox
- GN ReSound
- Hansaton
- Interton
- Oticon
- Philips
- Phonak
- Rexton
- Siemens
- Sonic Innovations
- Unitron
- Widex









Detailed, up-to-date compatibility lists of Phonak's MLx MicroLink receiver are available online at:

www.phonak.com/mlx

Phonak – Claro 211 Product Information at Page 1

FM features

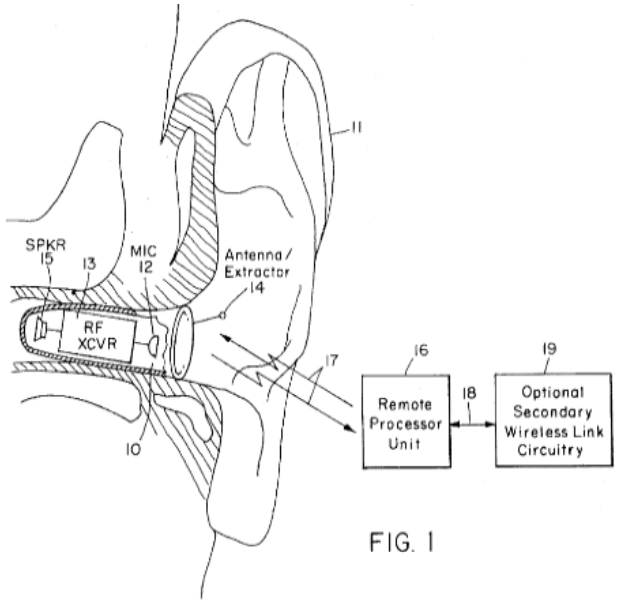
- World's smallest FM microchip
- Compatible with all Phonak FM transmitters
- Cordless TV and Radio link up
- FM performance corresponding to audiological fitting
- Soft Squelch system
- Built in antenna
- Band H (173–185 MHz); band J (186–201 MHz); band N (207–217 MHz)

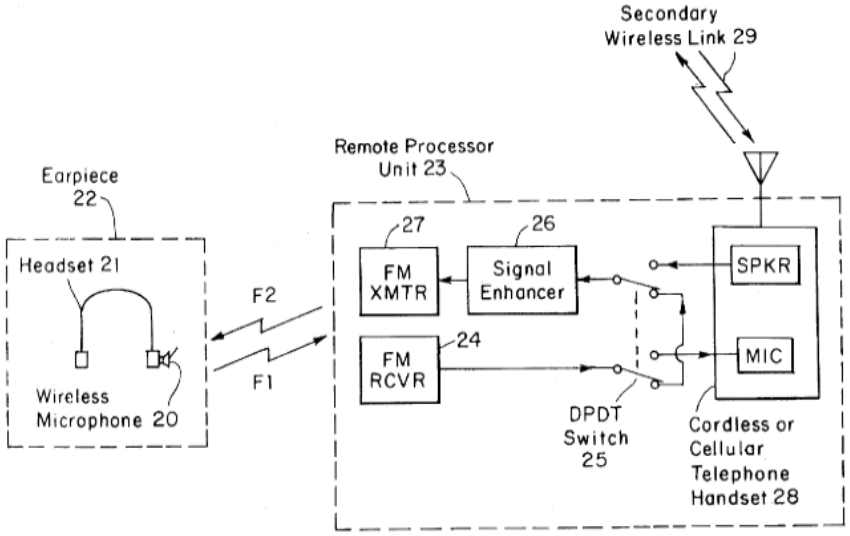
Phonak – Claro 211 Product Information at Page 1

Processing features

- Digital Perception Processing (DPP) in 20 critical bands
- Gain and output controlled by DPP
- 3 hearing programs (+ mute mode):
  - QuietAdapt for hearing in relative quiet
  - NoiseAdapt for hearing speech in noise
  - Program 3 telephone default (other choices available)
- FM programs: FM and FM+M
- AutoSelect: automatic switching between QuietAdapt and NoiseAdapt
- Claro's noise suppression technologies
  - Adaptive digital AudioZoom (dAZ)
  - Fine-scale Noise Canceler in 20 bands
  - Soft squelch

Claims	Prior Art
	Phonak – Claro 211 Product Information at Page 2 **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program. Phonak – FM Transmitters including Campus S and TelCom at Page 1 TelCom  TelCom (click to enlarge) Phonak's TelCom is the first stationary FM transmitter that **allows optimized television, telephone, and audio system usage**. While the user watches TV or listens to music, TelCom assures that the telephone will be heard when it rings. When the call is answered, TelCom automatically mutes the audio signal and transmits the caller's voice clearly and crisply. This does not affect the volume of the loudspeakers for others in the room. When the handset is replaced, the audio signal transmitted to the hearing instruments is turned back up to the previous volume setting. The TelCom has an operating range of 7 to 15 meters (23 to 50 feet). **For television, telephone, CD player, radio, personal computer and other equipment with audio output.** Oticon – DigiFocus White Paper (1996) at Page 11 Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time. Anderson '783 at Abstract A hearing aid or audio communication system includes an earpiece (10) that can be hidden in the ear canal, and which communicates wirelessly with a remote processor unit, or RPU (16), that enhances audio signals and can be concealed under clothing. Sounds from the environment are picked up by a microphone

Claims	Prior Art
	(12) in the earpiece and sent with other information over a two-way wireless link (17) to the RPU (16). The wireless link (17) uses microwaves for component miniaturization. Furthermore, use of radar technology to implement the wireless link (17), with an RPU (16) interrogator and earpiece (10) transponder, reduces earpiece size and power, as no microwave oscillator is needed in the earpiece (10). Optional secondary wireless link circuitry (19) can be used between the RPU (16) and a cellular telephone system or other sources of information. Electronic voice recognition and response can control system operation. Anderson '783 at FIG. 1  FIG. 1

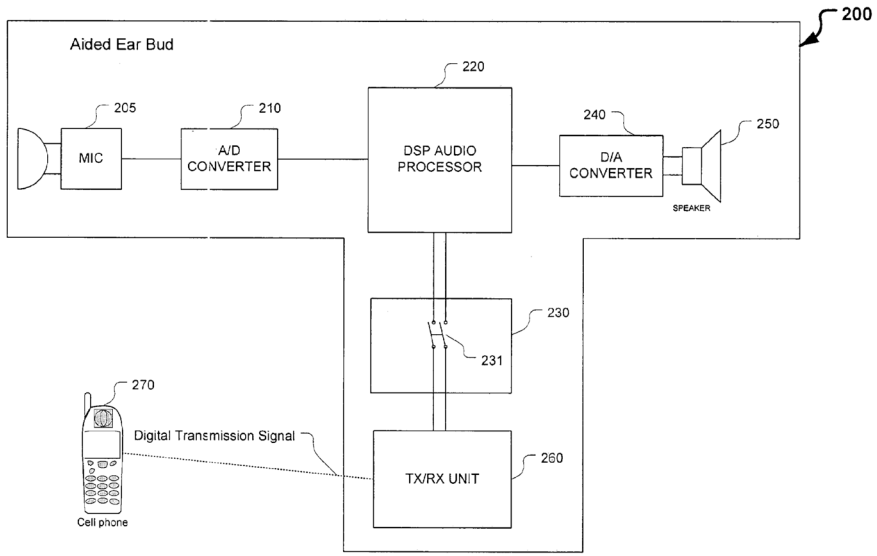
Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2 Anderson ’783 at 1:5–18 The present invention is directed to hearing aids, and in particular to a hearing aid having an earpiece housing worn in or at the ear and a remote processor unit (RPU) worn by or located near the user that wirelessly receives signals from and transmits signals to the earpiece. The present invention is also directed to the use of hearing aid systems as wireless communication devices for hands-free cellular telephone and mobile radio communication "handsets," covert operation and control of hearing aids, hearing protection and noise cancellation simultaneous with binaural hearing aid functions, hearing test equipment, location of misplaced hearing aid system components and wireless cochlear implants. Anderson ’783 at 1:50–65 The present invention uses an audio path that is different from the audio path used by devices in the prior art. The present invention uses a primary two-way wireless link (not a one-way link as in the prior art) between an earpiece worn in or at the user's ear and an RPU worn by or located near the user. In the present invention, audio signals from the environment are picked up by a microphone in the earpiece (not a hand-held microphone as used by wireless hearing aids in the prior art) and transmitted over the primary two-way wireless link to the RPU (instead of going to processor electronics contained in the earpiece, as

Claims	Prior Art
	in basic hearing aids known in the prior art), where the audio signals are enhanced according to the user's needs before transmission over the primary wireless link to the earpiece. Signal processing is performed in the RPU rather than the earpiece to take advantage of relaxed size and power constraints. Anderson ’783 at 2:20–39 It is an object of the present invention to provide a new and useful auditory aid for hearing-impaired persons (i.e., those having certain residual hearing) by removing audio signal enhancement functions from the earpiece and placing them in an RPU. Use of an RPU provides several advantages over systems that attempt to place all system capabilities within the earpiece. The RPU approach allows a simple earpiece design comprising a miniature low-power wireless transceiver, microphone and speaker. Note that the speaker is also known in the hearing aid art as a "receiver," but the term "speaker" is used here to avoid confusion (similarly, the term "talker," not "speaker," is used to describe a human producing vocal sounds). The resulting earpiece is extremely small, can be hidden from view in the ear canal if desired, and allows complete freedom of movement when a primary two-way wireless link to the RPU is used. Processing for all major system capabilities, such as amplification and other forms of signal enhancement, takes place in the RPU where size and power constraints are relaxed, leading to a cost-effective design. Anderson ’783 at 2:40–59 Another object of this invention is to provide supplemental audio information (e.g. a verbal warning from the RPU that an earpiece battery is low) and communication services (e.g., cellular telephone and paging services) to the user via the hearing aid system. Such services are accessed in a manner that can be made imperceptible to a casual observer if desired. The RPU (which contains a digital signal processor or other computer) acts as one source of information, e.g., by using a synthesized voice message to provide the time of day, and the RPU can also be used to access a secondary wireless link to the general subscriber telephone network or voice paging services. User control of hearing aid parameters and requests for information are accomplished using pushbuttons located on the RPU (including pushbuttons suitable for data entry in a covert manner) or voice recognition. Many sensors and peripheral devices can reside in or be attached to the RPU by wired or wireless means, and can provide a variety of information for different applications (e.g., heart pulse rate) as audio in the user's earpiece. Anderson ’783 at 2:60–3:5 A further object of this invention is to provide some degree of protection for the residual hearing capability of a hearing-impaired user (or a non-impaired user who wishes to avoid impairment) in a wireless hearing aid system having the simultaneous capabilities of noise cancellation and binaural

Claims	Prior Art
	processing (e.g., directionality). Other objects of the invention are to provide a convenient means for testing a user's hearing capability without the need for additional equipment, to assist users in the location of misplaced hearing aid system components, and to provide profoundly deaf cochlear implant patients with a wireless system allowing improved freedom of movement compared to existing wire-connected systems. Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative

Claims	Prior Art
	to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other

Claims	Prior Art
	information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 24:47–66 Morse code keys 956 or voice recognition capabilities implemented using the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. In this respect, the system acts as a "virtual handset" communication aid that replaces a conventional telephone handset in many applications (although users may additionally communicate with a conventional handset or speakerphone in the usual fashion). To accomplish the virtual handset function, the RPU 16 is equipped with optional secondary two-way wireless link circuitry 19 to communicate with a telephone or other communication network. The secondary wireless link is separate from, and does not interfere with, the primary wireless link 17 between the earpiece 10 and RPU 16. In applications where relatively high-power transmissions take place on the secondary link and the secondary link transmitter is integrated into the RPU, the RPU can be placed on a user's wrist or ankle away from vital organs to reduce any potential health risk relative to conventional wireless handsets. Anderson ’783 at 25:15–27 To obtain solicited information, user requests are processed in a variety of means well known in the art, including inputs via one or more pushbuttons on the RPU and voice recognition computer programs implemented in the RPU electronics. Voice recognition can be used almost exclusively, if desired, thereby eliminating nearly all RPU pushbutton controls. Voice commands recognized by the RPU can cause the

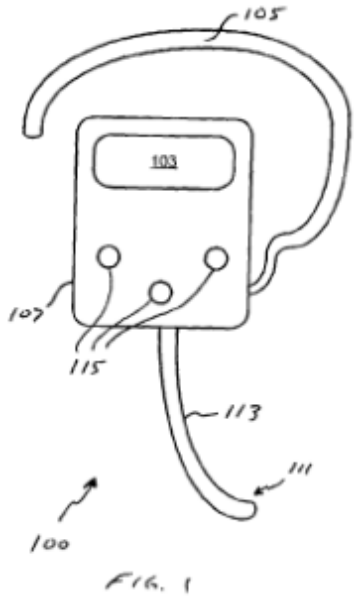
Claims	Prior Art
	RPU to initiate actions such as changing hearing aid parameter settings (e.g., volume), or activate appliances and environmental controls (e.g., aids for the physically handicapped) in a distributed wireless network via the secondary wireless link. Enzmann ’730 at FIG. 2  FIG. 2 Enzmann ’730 at 1:7–11 This invention relates to systems and methods for providing a hearing aid capable of communicating with a wireless phone. More particularly, this invention provides a hearing aid capable of simultaneously or alternatively performing the functions of an ear bud for the wireless phone. Enzmann ’730 at 2:17–23 Therefore, this invention provides systems and methods for adapting a hearing aid to operate as an ear bud device in addition to providing conventional hearing aid functions. Using this method, this invention is able to address many of the issues hearing impaired wireless phone users encounter while addressing the inherent problems associated with the conventional solutions.

Claims	Prior Art
	Enzmann ’730 at 3:55–57 In addition, the portability of the wireless phone is increased because the user's hearing device can now act as a wireless ear bud for the wireless phone. Enzmann ’730 at 4:14–16 In the second operating mode, the TX/RX unit 260 transmits and receives communications from the wireless phone 270. Enzmann ’730 at 4:43–50 The connection shown between the TX/RX unit 260 and the wireless phone 270 is preferably a wireless digital connection, such as Bluetooth, etc. However, it should be appreciated that the connection between the TX/RX unit 260 and the wireless phone 270 can be any kind of wireless or wired connection, and may be digital or analog, infrared (IR) or transmitted radio frequencies (RF) without, departing from the scope of the invention. Enzmann ’730 at 4:56–62 In the ear bud mode of operation, as mentioned above, the switch 231 is activated (i.e., closed), providing a continuous signal path for signals coming from the op amp 222 to be passed to the TX/RX unit 260 along signal line 226. Additionally, the TX/RX unit 260 can pass data to the op amp 223 along signal line 228 though the signal processing function 225 and signal line 228a. Enzmann ’730 at 4:63–5:5 Signals received at the microphone 205 are converted by the A/D converter 210, manipulated by the signal-processing function 221 which outputs 226a is then passed to the op amp 222. The output 226 and 226b of the op amp 222 are electronically connected to the TX/RX unit 260 and to the op amp 223. The output 226 connected to the TX/RX unit 260 is then output to the wireless phone 270. The signal from the TX/RX unit 260 is used by the wireless phone 270 as if the user was speaking into the microphone of the wireless phone. Enzmann ’730 at 5:13–19 The signals received at the microphone 205 are passed to the user as background noise and must be passed to the wireless phone 270 as if the user was speaking directly into the wireless phone 270. Remotely transmitting this signal to the wireless phone from the aided-ear bud 200 also allows the user to

Claims	Prior Art
	be located in a remote location away from the wireless phone 270. Enzmann ’730 at 5:20–30 The signal 226b passed to the op amp 223 is representative of ambient noise in the aided ear bud 200. This signal 226b is typically reduced in volume and combined with the input from the signal-processing function 225 to produce a mixed signal. The signal processing function 225 processes signal 228 from the TX/RX unit 260 and provides a second processing step so that the timing of the incoming signals from the microphone and the wireless phone are in sync and can be effectively combined to create a true representation of sound to the user. The op amp 223 outputs the mixed signal to the D/A converter 240, and finally to the speaker 250. Enzmann ’730 at 6:12–35 The amplification chart 325 contains a series of pre-stored dynamic frequency maps based on the user's hearing characteristics, the speaker characteristics and the ambient noise near the aided ear bud 200. The digital logic 320 will receive the transmitted signal from the TX/RX unit 260 and process those signals to an optimal form based on the user's hearing, speaker and ambient noise characteristics. The arrangement of the logic devices within the digital logic 320 will serve to produce each of the characteristics (wave shape) of the amplitude chart 325. The user's hearing characteristics remain static unless the aided ear bud 200 is reprogrammed to compensate for additional hearing loss or increase and different users. For example, as shown in FIG. 5, a mapping of a user's hearing characteristic (denoted by the “••” line) illustrates that the user lacks significant hearing capability in higher frequencies. The desired hearing characteristic is denoted by the “-••” line. This desired hearing characteristic depicts or represents how a non-hearing impaired person's hearing characteristics may be. As known to those skilled in the art, close representation of this hearing characteristic is achievable by adding a complementary characteristic to the user's hearing characteristic to produce a result that is very close to the desired characteristic. Each of the characteristics is corrected using the process shown in FIG. 5. Jannard ’526 at Abstract A wearable audio interface includes a support for positioning the plurality of speakers juxtaposed to and spaced from the ears of a wearer. The audio device can include wireless networking electronics so as to allow the device to interact with other wireless network devices.

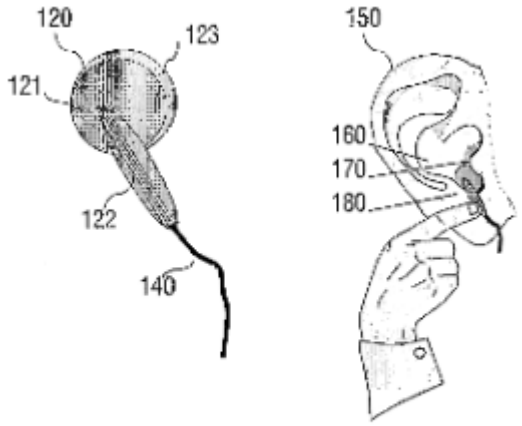
Claims	Prior Art
	Jannard ’526 at 2:10–13 The system also comprises source electronics electrically coupled with the receiver electronics and configured to wirelessly transmit information to the receiver electronics. Jannard ’526 at 19:41–45 As illustrated in FIG. 8, the audio device 10C can be worn on the head 18 of a user U. Preferably, the audio device 10C is configured to provide one or two-way wireless communication with a source device, or the source device can be incorporated into the audio device 10C. Jannard ’526 at 19:48–54 The user U can carry a “body borne” source device B such as, for example, but without limitation, a cellular phone, an MP3 player, a “two-way” radio, a palmtop computer, or a laptop computer. As such, the user U can use the audio device 10C to receive and listen to audio signals from the source device B, and/or transmit audio signals to the source device B. Jannard ’526 at 20:10–25 In general, the component configuration of FIG. 8 enables the audio device 10C to carry interface electronics with the user, such as audio output and audio input. However, the source electronics such as the MP3 player, cellular phone, computer or the like may be off board, or located remotely from the audio device 10C. This enables the audio device 10C to accomplish complex electronic functions, while retaining a sleek, low weight configuration. Thus, the audio device 10C is in communication with the off board source electronics device B. The off board source device B may be located anywhere within the working range of the audio device 10C. In many applications, the source electronics B will be carried by the wearer, such as on a belt clip, pocket, purse, backpack, integrated with “smart” clothing, or the like. This accomplishes the function of off loading the bulk and weight of the source electronics from the headset. Jannard ’526 at 25:21–37 The audio device 10C can carry any of a variety of receivers and/or transmitters, such as transceiver 114. For example, but without limitation, the audio device 10C can carry receivers and/or transmitters for music or for global positioning. In another example, the audio device 10C can carry receivers and/or transmitters for telecommunications (i.e., telecommunications devices). As used herein, the term “telecommunications devices” is intended to include telephone components as well as devices for

Claims	Prior Art
	communicating with a telephone. For example, “telecommunications devices” can include one or more transceivers for transmitting an audio signal to a cellular phone to be transmitted by the cellular phone as the speaker's voice, and/or for receiving an audio signal from a cellular phone representing a caller's voice. Of course, other audio, video, or data signals can be transmitted between the audio device 10C and such a cellular phone through such transceivers. Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. Jannard ’526 at 30:40–52 The headset profile is designed to be used for interfacing a headset having one earphone, a microphone, and a transceiver worn by the wearer, for example, on a belt clip, with a cordless phone through a wireless connection. According to the headset profile, certain commands can be issued from a headset, such as the audio devices 10, 10A, 10A′, 10B, 10C, 10D, and 10E, using an AT command protocol. In such a protocol, text commands must be input to the BLUETOOTH™ device, which the BLUETOOTH™

Claims	Prior Art
	device then transmits wirelessly to a synchronized BLUETOOTH™ enabled device. Such commands include, for example, but without limitation, initiating a call, terminating a call, and redialing a previously dialed number. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[3a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a touch sensitive surface at least partially contained within the head-worn personal electronic apparatus and configured to sense a touch input associated with the user,	DHA systems, alone or in view of Bodley or Hollemans '490 or Seshadri '839, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a touch sensitive surface at least partially contained within the head-worn personal electronic apparatus and configured to sense a touch input associated with the user." Bodley '433 at FIG. 1  FIG. 1 Bodley '433 at 2:7–8 FIG. 1 is a frontal view of a generic headset that includes a display in accordance with the invention;

Claims	Prior Art
	Bodley ’433 at 2:38–50 Although headset 100 may use any of a variety of means of attaching to a user (e.g., headband, earpiece, etc.) and thus allowing hands-free operation of the headset, preferably headset 100 includes an earpiece 105 (e.g., an ear hook) that is coupled to the user's ear, or a portion thereof. More preferably, earpiece 105 fits around a portion of the user's ear. Earpiece 105 is coupled to body 107 of the headset, body 107 housing the various electronic components necessary to provide functionality to headset 100. Attached to the back surface of body 107 is an output transducer 109 (e.g., a speaker). Also coupled to body 107 is an input transducer 111 (e.g., a microphone), which, in this embodiment, is located at the end of a boom member 113. Bodley ’433 at 2:65–3:22 Display 103 can be used to provide the user with various types of information depending upon the design goals for the specific headset. The design goals are primarily based on the expected headset use, desired headset functionality, display size, display resolution, allowable headset power drain, and the processing capabilities of the headset on-board processor (also referred to as a controller or an interface controller). Examples of information that can be displayed on display 103 are headset battery level, headset signal level, call status (e.g., dialing, in use, etc.), caller identification, current time/date, and timer information such as elapsed time associated with a particular call. Display 103 can also be configured to flash or otherwise visually indicate an incoming call. Assuming that the information is provided to the headset processor, display 103 can also display the battery level of the base unit (e.g., cell phone, dongle, etc.) and the signal strength of the base unit (e.g., cell phone). Additionally, display 103 can be used during headset configuration (e.g., to adjust or set the ring mode, ring loudness, headset volume, display mode, voice dialing, time, date, etc.) or to provide additional headset functionality (e.g., phone lists, text messages, dialing using virtual numeric keypad or phone list dialing). Additionally, display 103 can be touch sensitive, thus providing a touch screen for data input (e.g., phone numbers into a phone list, dialing using a virtual numeric keypad). Hollemans ’490 at 2:6–9 This object is realized in that the controller has an outer surface with a touch-sensitive area, the controller being arranged to be substantially worn in or by a human ear, the controller being further arranged to detect the touch-sensitive area being touched, and to send the control signal in response to detecting the touch-sensitive area being touched.

Claims	Prior Art
	Hollemans ’490 at 2:10–18 The remote controller thus gets a fixed position with respect to the user in or by the human ear. It is easy to find one's ear. This prevents looking for the remote controller. In addition, the user is not tempted to look at the remote controller, because there is nothing relevant to be seen for controlling the device. To avoid having to find tiny buttons, merely touching the remote controller somewhere on its touch-sensitive area operates said remote controller, which detects being touched and subsequently sends the control signal to the device. Also the problem of finding the proper orientation of the remote controller is solved by wearing the remote controller in or by the ear, because its orientation becomes fixed with respect to the user. Hollemans ’490 at 2:25–33 Advantageously, the controller is arranged to detect a temporal pattern in the touch-sensitive area being touched, and to send the control signal in response to detecting the temporal pattern. In general, the device offers a plurality of functions and capabilities. Rather than having more buttons on the remote control that may be hard to distinguish, the single touch-sensitive area is used for controlling the plurality of functions. The temporal pattern is a particularly appropriate user interface, because it is easy to create temporal patterns in touching the area rhythmically and because the temporal patterns can constitute a natural and consistent interface. An example is a single short tap for pause/play, double short tap for next track, triple short tap for next artist/album and a long tap for adjusting the volume. Hollemans ’490 at 3:1–6 Advantageously, the outer surface has a further touch-sensitive area, such that the further touch-sensitive area is touched substantially by the ear when the controller is substantially worn in or by a human ear, the controller being arranged to send the control signal only if the further touch-sensitive area is touched. This will prevent the accidental remote control of the device when the remote controller is not worn, thus preventing unwanted activation of the device and unnecessary battery exhaustion.

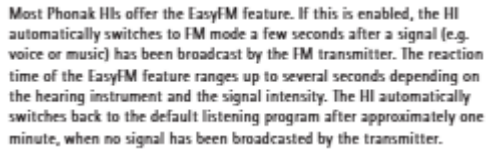
Claims	Prior Art
	Hollemans ’490 at FIG. 2  FIG.2 Seshadri ’839 at [0041] Wearable earpiece 12, as shown in FIG. 5, includes antenna 46, transmit/receive switch 122, receiver module 40, data recovery module 42, speaker module 44, transmitter module 120, input module 128 and display module 132. Receiver module 40, data recovery module 42 and speaker module 44 operate as discussed with reference to FIG. 4, with the exception of the data recovery module 42, to produce display information that is provided to display module 132. For instance, the received RF signal may include display information such as caller ID, command information, etc. which is separated by data recovery module 42 and provided to display module 132, which may be an LCD display, plasma display, etc. Seshadri ’839 at [0042] Input module 128, which may be a keypad, touch screen, voice recognition circuit, or other like user interfaces, receives user commands and produces digital command messages 124 there from. Such digital command messages 124 includes, but are not limited to, packet size, synchronization information, frequency hopping initiation information, timeslot allocation information, link establishment information, piconet address information, fast-forward, play, pause, volume adjust, record, stop and rewind. A POSITA would have been motivated to combine the teachings of DHA systems with Bodley or Hollemans or Seshadri to meet the claim limitations with reasonable expectation of success. DHA systems discloses that

Claims	Prior Art
	an aided ear bud can operate according to characteristics that are device and user specific and that can be set using information stored in the memory of the earbud. However, the extent DHA systems does not disclose details for setting or adjusting those characteristics, a POSITA would recognize Bodley or Hollemans or Seshadri to provide a solution to this problem: the ear bud can be provided with a touch sensitive surface for setting and adjusting various characteristics of the ear bud. A POSITA would be motivated to look to Bodley or Hollemans or Seshadri because DHA systems and these references share various structural and functional similarities that render the inventions readily combinable. For example, both DHA systems and Bodley disclose ear-worn devices for providing audio output to a user. Enzmann, FIG. 2; Bodley, FIG. 1. Each include similar components, such as microphones, speakers, wireless transceivers, and control circuitry. Bodley, FIG. 8. Accordingly, combining the teachings of DHA systems and Bodley or Hollemans or Seshadri would involve combining known components to yield predictable results. To the extent that DHA systems and Bodley or Hollemans ’490 or Seshadri ’839do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[3b] wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus.	DHA systems, alone or in view of Bodley or Hollemans ’490 or Seshadri ’839, expressly or inherently disclose “wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus.” Phonak – Fitting Guide for MLxi and MLxi Baha at Page 3 Most Phonak HIs offer the EasyFM feature. If this is enabled, the HI automatically switches to FM mode a few seconds after a signal (e.g. voice or music) has been broadcast by the FM transmitter. The reaction time of the EasyFM feature ranges up to several seconds depending on the hearing instrument and the signal intensity. The HI automatically switches back to the default listening program after approximately one minute, when no signal has been broadcasted by the transmitter. Bodley ’433 at 2:65–3:22 Display 103 can be used to provide the user with various types of information depending upon the design goals for the specific headset. The design goals are primarily based on the expected headset use, desired headset functionality, display size, display resolution, allowable headset power drain, and the processing capabilities of the headset on-board processor (also referred to as a controller or an interface controller). Examples of information that can be displayed on display 103 are headset battery level, headset signal level, call status (e.g., dialing, in use, etc.), caller identification, current time/date, and timer information such as elapsed time associated with a particular call. Display 103 can also be configured to flash or otherwise visually indicate an incoming call. Assuming that the information is provided to the headset

Claims	Prior Art
	processor, display 103 can also display the battery level of the base unit (e.g., cell phone, dongle, etc.) and the signal strength of the base unit (e.g., cell phone). Additionally, display 103 can be used during headset configuration (e.g., to adjust or set the ring mode, ring loudness, headset volume, display mode, voice dialing, time, date, etc.) or to provide additional headset functionality (e.g., phone lists, text messages, dialing using virtual numeric keypad or phone list dialing). Additionally, display 103 can be touch sensitive, thus providing a touch screen for data input (e.g., phone numbers into a phone list, dialing using a virtual numeric keypad). Hollemans ’490 at 2:10–18 The remote controller thus gets a fixed position with respect to the user in or by the human ear. It is easy to find one's ear. This prevents looking for the remote controller. In addition, the user is not tempted to look at the remote controller, because there is nothing relevant to be seen for controlling the device. To avoid having to find tiny buttons, merely touching the remote controller somewhere on its touch-sensitive area operates said remote controller, which detects being touched and subsequently sends the control signal to the device. Also the problem of finding the proper orientation of the remote controller is solved by wearing the remote controller in or by the ear, because its orientation becomes fixed with respect to the user. Hollemans ’490 at 2:25–33 Advantageously, the controller is arranged to detect a temporal pattern in the touch-sensitive area being touched, and to send the control signal in response to detecting the temporal pattern. In general, the device offers a plurality of functions and capabilities. Rather than having more buttons on the remote control that may be hard to distinguish, the single touch-sensitive area is used for controlling the plurality of functions. The temporal pattern is a particularly appropriate user interface, because it is easy to create temporal patterns in touching the area rhythmically and because the temporal patterns can constitute a natural and consistent interface. An example is a single short tap for pause/play, double short tap for next track, triple short tap for next artist/album and a long tap for adjusting the volume. Hollemans ’490 at 3:1–6 Advantageously, the outer surface has a further touch-sensitive area, such that the further touch-sensitive area is touched substantially by the ear when the controller is substantially worn in or by a human ear, the controller being arranged to send the control signal only if the further touch-sensitive area is touched. This will prevent the accidental remote control of the device when the remote controller is not worn, thus

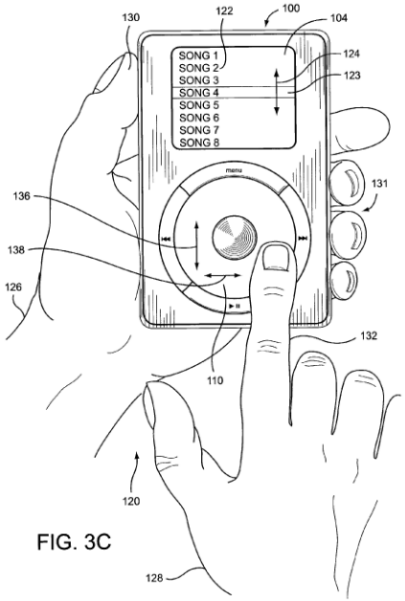
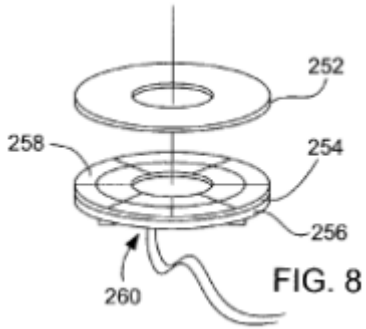
Claims	Prior Art
	preventing unwanted activation of the device and unnecessary battery exhaustion. Hollemans ’490 at 3:7–10 Advantageously, the controller is arranged to send a further control signal to the device if the further touch-sensitive area is touched. This makes it, for example, possible for the device to pause playing music as soon as the earpiece is taken out and to resume playing music upon re-insertion. Hollemans ’490 at 3:11–21 In an embodiment of the personal audio system according to the invention, the system comprises a second controller for remotely controlling the device by sending a further control signal to the device, the second controller having an outer surface with a further touch-sensitive area, the second controller being arranged to be substantially worn in or by a human ear, and the second controller being further arranged to detect a further temporal pattern in the further touch-sensitive area being touched, and to send the further control signal in response to detecting the further temporal pattern. A second controller fits in with the natural symmetry of the human head. It also considerably enhances the user interface for controlling the device. For example, the symmetry can be exploited in such a way that a long tap at the left remote controller decreases the volume, but a long tap at the right controller increases the volume. Hollemans ’490 at FIG. 3  FIG.3

Claims	Prior Art
	Hollemans ’490 at FIG. 4  FIG.4 Seshadri ’839 also discloses this feature. Seshadri ’839 at [0042] Input module 128, which may be a keypad, touch screen, voice recognition circuit, or other like user interfaces, receives user commands and produces digital command messages 124 there from. Such digital command messages 124 includes, but are not limited to, packet size, synchronization information, frequency hopping initiation information, timeslot allocation information, link establishment information, piconet address information, fast-forward, play, pause, volume adjust, record, stop and rewind. A POSITA would have been motivated to combine the teachings of DHA systems and Bodley or Hollemans ’490 or Seshadri ’839, with a reasonable expectation of success, for the reasons described above with respect to claim 3a. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
5. A head-worn personal electronic apparatus recited in claim 3, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user.	DHA systems, alone or in view of Bodley or Hollemans ’490, expressly or inherently disclose “[a] head-worn personal electronic apparatus recited in claim 3, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user.”

Claims	Prior Art
	Phonak – Fitting Guide for MLxi and MLxi Baha at Page 3  Most Phonak HIs offer the EasyFM feature. If this is enabled, the HI automatically switches to FM mode a few seconds after a signal (e.g. voice or music) has been broadcast by the FM transmitter. The reaction time of the EasyFM feature ranges up to several seconds depending on the hearing instrument and the signal intensity. The HI automatically switches back to the default listening program after approximately one minute, when no signal has been broadcasted by the transmitter. Bodley ’433 at 2:65–3:22 Display 103 can be used to provide the user with various types of information depending upon the design goals for the specific headset. The design goals are primarily based on the expected headset use, desired headset functionality, display size, display resolution, allowable headset power drain, and the processing capabilities of the headset on-board processor (also referred to as a controller or an interface controller). Examples of information that can be displayed on display 103 are headset battery level, headset signal level, call status (e.g., dialing, in use, etc.), caller identification, current time/date, and timer information such as elapsed time associated with a particular call. Display 103 can also be configured to flash or otherwise visually indicate an incoming call. Assuming that the information is provided to the headset processor, display 103 can also display the battery level of the base unit (e.g., cell phone, dongle, etc.) and the signal strength of the base unit (e.g., cell phone). Additionally, display 103 can be used during headset configuration (e.g., to adjust or set the ring mode, ring loudness, headset volume, display mode, voice dialing, time, date, etc.) or to provide additional headset functionality (e.g., phone lists, text messages, dialing using virtual numeric keypad or phone list dialing). Additionally, display 103 can be touch sensitive, thus providing a touch screen for data input (e.g., phone numbers into a phone list, dialing using a virtual numeric keypad). Hollemans ’490 at 2:10–18 The remote controller thus gets a fixed position with respect to the user in or by the human ear. It is easy to find one's ear. This prevents looking for the remote controller. In addition, the user is not tempted to look at the remote controller, because there is nothing relevant to be seen for controlling the device. To avoid having to find tiny buttons, merely touching the remote controller somewhere on its touch-sensitive area operates said remote controller, which detects being touched and subsequently sends the control signal to the device. Also the problem of finding the proper orientation of the remote controller is solved by wearing the remote controller in or by the ear, because its orientation becomes fixed with respect to the user.

Claims	Prior Art
	Hollemans ’490 at 2:25–33 Advantageously, the controller is arranged to detect a temporal pattern in the touch-sensitive area being touched, and to send the control signal in response to detecting the temporal pattern. In general, the device offers a plurality of functions and capabilities. Rather than having more buttons on the remote control that may be hard to distinguish, the single touch-sensitive area is used for controlling the plurality of functions. The temporal pattern is a particularly appropriate user interface, because it is easy to create temporal patterns in touching the area rhythmically and because the temporal patterns can constitute a natural and consistent interface. An example is a single short tap for pause/play, double short tap for next track, triple short tap for next artist/album and a long tap for adjusting the volume. Hollemans ’490 at 3:1–6 Advantageously, the outer surface has a further touch-sensitive area, such that the further touch-sensitive area is touched substantially by the ear when the controller is substantially worn in or by a human ear, the controller being arranged to send the control signal only if the further touch-sensitive area is touched. This will prevent the accidental remote control of the device when the remote controller is not worn, thus preventing unwanted activation of the device and unnecessary battery exhaustion. Hollemans ’490 at 3:7–10 Advantageously, the controller is arranged to send a further control signal to the device if the further touch-sensitive area is touched. This makes it, for example, possible for the device to pause playing music as soon as the earpiece is taken out and to resume playing music upon re-insertion. Hollemans ’490 at 3:11–21 In an embodiment of the personal audio system according to the invention, the system comprises a second controller for remotely controlling the device by sending a further control signal to the device, the second controller having an outer surface with a further touch-sensitive area, the second controller being arranged to be substantially worn in or by a human ear, and the second controller being further arranged to detect a further temporal pattern in the further touch-sensitive area being touched, and to send the further control signal in response to detecting the further temporal pattern. A second controller fits in with the natural symmetry of the human head. It also considerably enhances the user interface for controlling the device. For example, the symmetry can be exploited in such a way that a long tap at the left remote controller decreases the volume, but a long tap at the right controller increases the volume.

Claims	Prior Art
	Hollemans ’490 at FIG. 3  FIG.3 Hollemans ’490 at FIG. 4  FIG.4 A POSITA would have been motivated to combine the teachings of DHA systems and Bodley or Hollemans, with a reasonable expectation of success, for the reasons described above with respect to claim 3. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
6. A head-worn personal electronic apparatus as recited in claim 3, wherein the head-worn personal electronic apparatus is configured to detect at least a sliding of a	DHA systems in view of Zadesky or Hollemans expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 3, wherein the head-worn personal electronic apparatus is configured to detect at least a sliding of a finger action with respect to the touch sensitive surface.”

Claims	Prior Art
finger action with respect to the touch sensitive surface.	Zadesky ’230 at FIG. 3C  FIG. 3C Zadesky ’230 at FIG. 8  FIG. 8 Zadesky ’230 at 7:28–44 The touch pad generally consists of a touchable outer surface 111 for receiving a finger for manipulation on the touch pad 110. Although not shown in FIG. 2, beneath the touchable outer surface 111 is a sensor arrangement. The sensor arrangement includes a plurality of sensors that are configured to activate as the finger passes over them. In the simplest case, an electrical signal is produced each time the finger passes a sensor. The number of signals in a given time frame may indicate location, direction, speed and

Claims	Prior Art
	acceleration of the finger on the touch pad, i.e., the more signals, the more the user moved his or her finger. In most cases, the signals are monitored by an electronic interface that converts the number, combination and frequency of the signals into location, direction, speed and acceleration information. This information may then be used by the media player 100 to perform the desired control function on the display screen 104. Zadesky '230 at 8:52–67 FIGS. 3A-3C show the media player 100 of FIG. 2 being used by a user 120, in accordance with different embodiments of the invention. In all of these embodiments, the user 120 is linearly scrolling 104 (as shown by arrow 124) through a list of songs 122 displayed on the display screen via a slider bar 123. As shown, the media device 100 is comfortably held by one hand 126 while being comfortably addressed by the other hand 128. This configuration generally allows the user 120 to easily actuate the touch pad 110 with one or more fingers. For example, the thumb 130 and rightmost fingers 131 (or leftmost fingers if left handed) of the first hand 126 are used to grip the sides of the media player 100 while a finger 132 of the opposite hand 128 is used to actuate the touch pad 110. As shown, the entire top surface of the touch pad 110 is accessible to the user's finger 130. Zadesky '230 at 9:16–21 Referring to FIG. 3C, and in accordance with one embodiment of the invention, the user 120 can slide his or her finger 132 substantially tangentially from all sides of the touch pad 110. For example, the touch pad 110 may be actuated forwards and backwards as shown by arrows 136 and side to side by arrows 138. Zadesky '230 at 11:50–12:20 FIG. 8 is a partially broken away perspective view of an annular capacitive touch pad 250, in accordance with one embodiment of the present invention. By way of example, the annular capacitive touch pad 250 may correspond to the touch pad of FIG. 2. The annular capacitive touch pad 250 is arranged to detect changes in capacitance as the user swirls an object such as a finger around the touch pad 250. The annular capacitive touch pad 250 is also arranged to detect changes in capacitance as the user moves their finger radially across the touch pad 250. The annular capacitive touch pad 250 is formed from various layers including at least a label layer 252, an electrode layer 254 and a circuit board 256. The label layer 252 is disposed over the electrode layer 254 and the electrode layer 254 is disposed over the circuit board 256. At least the label 252 and electrode layer 254 are annular such that they are defined by concentric circles, i.e., they have an inner perimeter and an outer perimeter. The circuit board 256 is generally a circular piece having an outer perimeter that coincides with the outer perimeter of the label 252 and electrode

Claims	Prior Art
	layer 254. It should be noted, however, that in some cases the circuit board 256 may be annular or the label 252 and electrode layer 254 may be circular. The label layer 252 serves to protect the underlayers and to provide a surface for allowing a finger to slide thereon. The surface is generally smooth so that the finger does not stick to it when moved. The label layer 252 also provides an insulating layer between the finger and the electrode layer 254. The electrode layer 254 includes a plurality of spatially distinct electrodes 258 that have positions based on the polar coordinate system. For instance, the electrodes 258 are positioned both angularly and radially on the circuit board 256 such that each of the electrodes 258 defines a distinct angular and radial position thereon. Any suitable number of electrodes 258 may be used. In most cases, it would be desirable to increase the number of electrodes 258 so as to provide higher resolution, i.e., more information can be used for things such as acceleration. Hollemans '490 at 8:19–22 In a third way, the touch-sensitive area is used for picking up a gesture made on the area with e.g. a stylus or a finger. The remote control may be extended for this purpose with means for detecting a gesture, e.g. a laser beetle, a small camera, an IR movement sensor, or a cross-capacitance sensor. Hollemans '490 at 8:30–33 The gesture detected may be a user defined gesture for activating a user defined function. The gesture detected may be used for user identification by matching with initials or an autograph. The gesture may extend into two dimensions, e.g. a planar gesture, or it may extend into three dimensions. A POSITA would have been motivated to combine the teachings of DHA systems and Zadesky or Hollemans to meet the claim limitations with reasonable expectation of success. As discussed above with respect to claim 3, DHA systems discloses that an aided ear bud can operate according to characteristics that are device and user specific and that can be set using information stored in the memory of the earbud, but it does not expressly include specific details of how those characteristics can be set. A POSITA would recognize that Bodley provides a solution to this problem—the ear bud can be provided with a touch sensitive surface that can be used for setting and adjusting various characteristics for the earpiece. A POSITA would be motivated to look to Bodley because DHA systems and Bodley both disclose ear-worn devices for providing audio output to a user and share similar components, such as microphones, speakers, wireless transceivers, and control circuitry. Moreover, the POSITA would be motivated to use Zadesky's capacitive sensing touch pad in the modified DHA systems. Bodley already discloses that a display for controlling settings and functions of an earpiece can include a touch sensitive surface. Bodley, 3:20-23. However, the specific details of the functionality of the touch sensitive surface are not expressly disclosed in Bodley. A POSITA thus would have

Claims	Prior Art
	been motivated to look to other references to supply these missing details. A POSITA would have been motivated to look to Zadesky at least because it is in the same general field of endeavor as DHA systems and Bodley. As discussed above, both DHA systems and Bodley disclose head-worn devices for providing audio output to a user. Zadesky discloses a media player that is "dedicated to processing media such as audio" and "allow[s] a user to listen to music." Zadesky, 4:62-67. Zadesky is therefore from the same general field of endeavor as Enzmann and Bodley—providing audio output to a user. Thus, the POSITA would have been motivated to combine the teachings of these references with a reasonable expectation of success. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[7a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user; and	DHA systems, alone or in view of Anderson or Jannard or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user." Phonak – Perseo Brochure at Page 23 

Claims	Prior Art
	Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user. Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

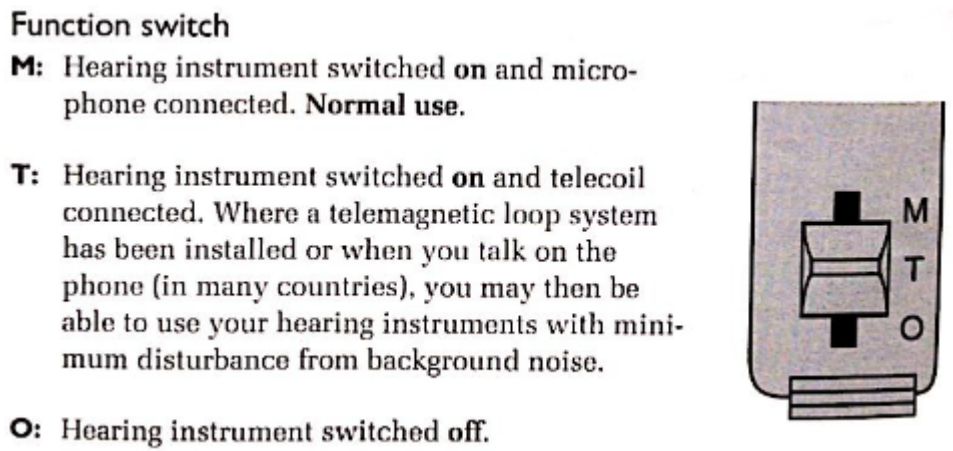
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

7

Oticon – DigiFocus Compact User Instructional Brochure at Page 4



Function switch

M: Hearing instrument switched **on** and microphone connected. **Normal use.**

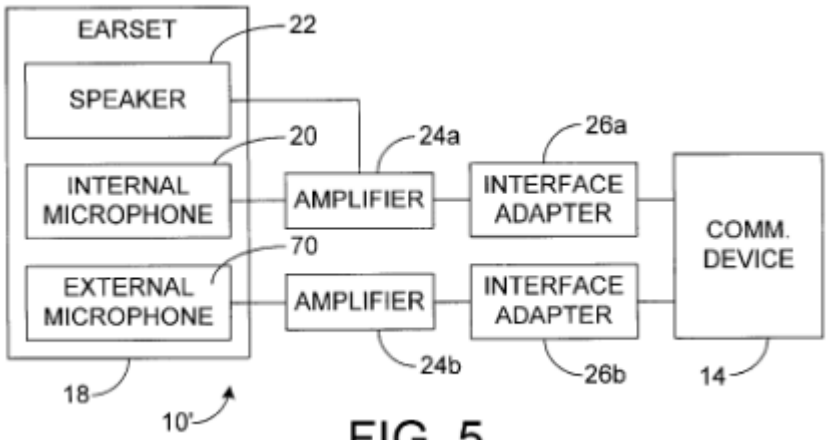
T: Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise.

O: Hearing instrument switched **off.**

Anderson '783 at 26:54–65

Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece

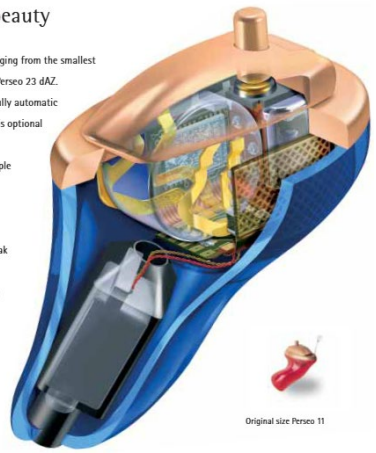
Claims	Prior Art
	microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Jannard ’526 at 2:26–34 In accordance with another embodiment of at least one of the inventions disclosed herein, an eyeglass frame comprises a support for supporting at least one lens in the path of a wearer's field of view, a first ear stem attached to the support, a second ear stem attached to the support, and at least one microphone supported by at least one of the support, first ear stem, and second ear stem. The microphone is advantageously arranged to face towards the head of a wearer of the eyeglass frame. Jannard ’526 at 18:23–26 Further, the audio device 10A″ can include noise cancellation electronics (not shown) configured to filter wind-generated noise from an audio signal transmitted from the microphone 75. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be

Claims	Prior Art
	supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or

Claims	Prior Art
	wireless, telephone from an RF transceiver of the telephone. Nemirovski ’379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. To the extend not disclosed by DHA systems, Nemirovski discloses the details of the claimed feature. As disclosed in the above–cited citations, Nemirovski ’379 discloses a first microphone for picking up audio from a user (the internal microphone), a second microphone for picking up audio generated outward away from the user (the external microphone), and that the audio is used for noise cancelation. A POSITA would thus have understood that the audio picked up by Nemirovski ’379’s second (external) microphone is used for noise cancelation. A POSITA would have been motivated to combine the DHA system and Nemirovski ’379 to meet the claim limitations with reasonable expectation of success for at least the reasons described above in reference to claim [1pre]. The DHA system discloses that its hearing aid system can performs noise cancellation to enhance speech and Nemirovski ’379 discloses one such example of a known technique for producing “a relatively noise-free speech signal” that includes “cancel[ing] externally generated noise from

Claims	Prior Art
	the detections of the internal microphone." See id. at 10:11-14, 24-25. It therefore would have been obvious to a POSITA to employ Nemirovski '379's noise cancellation techniques in the DHA system. The features of Nemirovski' 379's earset would have been readily combinable with the Phonak hearing aid by a POSITA with a reasonable expectation of success. Both systems disclose ear-worn devices for use with a communication system and can also include noise canceling functionality. Both devices also share other analogous and familiar computer components, such as microphones, speakers, and amplifiers. Accordingly, combining their respective teachings would involve combining known components to yield predictable results. To the extent that the DHA system, alone or in view of Anderson, Jannard, or Nemirovski, does not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[7b] a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user.	DHA systems, alone or in view of Anderson or Jannard or Nemirovski, expressly or inherently disclose "a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user." Phonak – Perseo Brochure at Page 7 **Convenience and Comfort** Perseo is easier to wear and easier to fit ■ For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. ■ This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. ■ Perseo is easily programmed for individual needs using the **Listening Situation Manager**. **Clarity** Enhanced performance in multiple listening situations ■ The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. ■ Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. ■ For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. ■ Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 11 ■ With **dual-base automatic** the fitter has direct access to the two separate base-programs. Performance can be matched to each user's needs in quiet situations without disrupting performance in noisy situations and vice versa. The result is less fine-tuning effort than with single-base instruments. With **single-base automatic** time is wasted "juggling" a single set of fitting parameters while attempting to balance performance in multiple listening situations. ■ To further simplify fitting, PersonalLogic is pre-set according to typical needs and is easily re-programmed to individual preferences using the **Listening Situation Manager (LSM)**. Ensuring that an automatic hearing system responds correctly in noisy situations where there is no speech is an important consideration for safety and to eliminate feelings of isolation. The **LSM** provides this functionality with a single mouse click. Matching the system's sensitivity to noise with that of the client's, and controlling the auditory contrast between quiet and noisy settings is equally easy. Phonak – Perseo Brochure at Page 13 Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer.

Claims	Prior Art
	Phonak – Perseo Brochure at p. 14 **Rapid-tracking dAZ** Phonak pioneered the development of directional microphone systems. Not surprisingly, Perseo **dAZ** with new soft switching brings even better performance. Perseo **dAZ** is automatically activated in appropriate situations and immediately reduces noise from beside and behind the wearer. The wearer experiences an immediate reduction of background noise and enhanced speech clarity. Rapid tracking ensures that even fast moving noise sources are controlled. Phonak – Perseo Brochure at Page 22 The beauty of high technology Perseo® ITEs – hidden beauty The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs. In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.  Original size Perseo 11

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
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Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

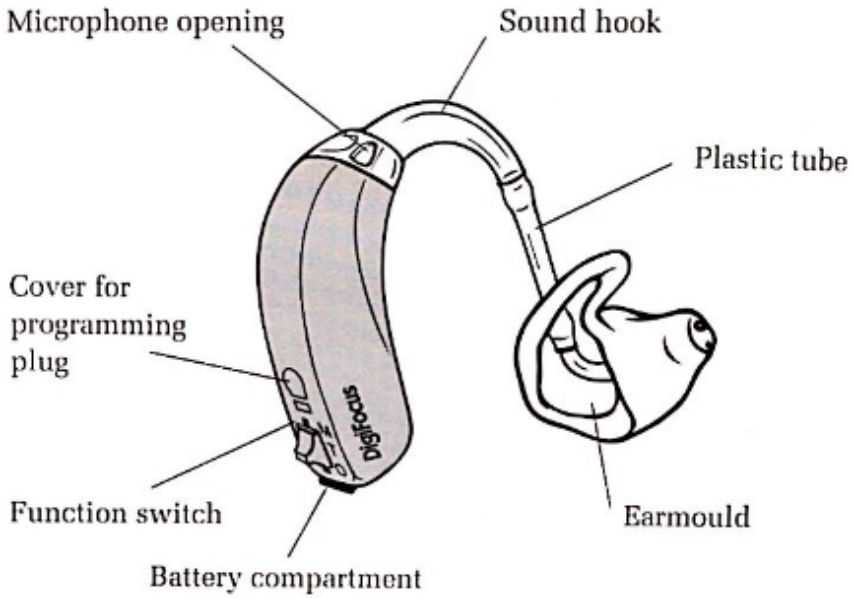
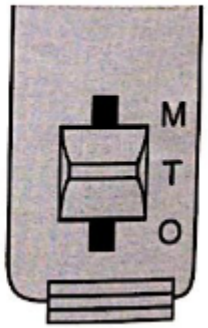
Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

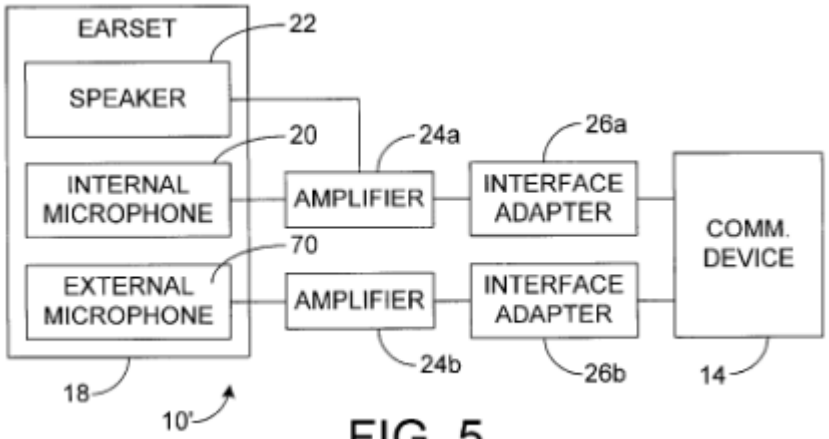
Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

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Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Oticon – DigiFocus Compact User Instructional Brochure at Page 4 **Function switch** **M:** Hearing instrument switched **on** and microphone connected. **Normal use.** **T:** Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise. **O:** Hearing instrument switched **off.**  Anderson '783 at 23:65–24:14 Background noise and competing talker cancellation are achieved using any of a number of techniques and circuits well known in the art, e.g., variations of the Least Mean Squares (LMS) technique as

Claims	Prior Art
	implemented in the Motorola DSP56200 LMS filter integrated circuit, or the equivalent function implemented in the general-purpose RPU DSP 948. Note that for all binaural signal processing functions, data contained in replies from the left earpiece (i.e., audio data from the left earpiece microphone) is processed in the RPU DSP 948 by combining said data with other data contained in replies from the right earpiece (i.e., audio data from the right earpiece microphone). The processed audio data transmitted from the RPU to the left and right earpiece speakers via RPU interrogations is therefore a function of data received from both the left and right earpiece microphones via earpiece replies. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes.

Claims	Prior Art
	Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone. Nemirovski ’379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from

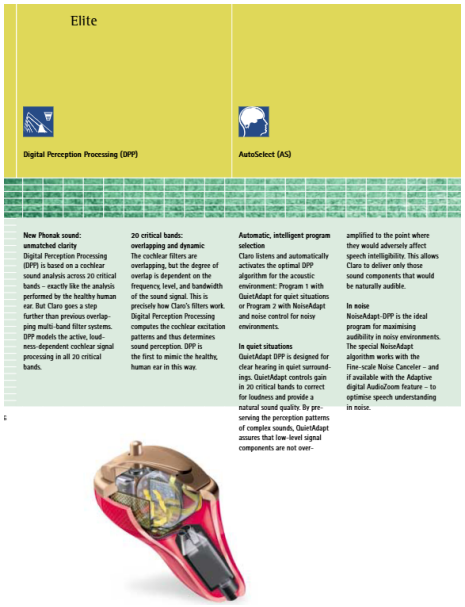
Claims	Prior Art
	the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. Nemirovski ’379 at 9:33–39 The external microphone 70 is disposed on a exterior portion 72 of the housing 30 and is pointed away from the wearer of the earset 18 assembly. The external microphone 70 can be secured to the housing 30 with adhesive and can be disposed in a recess defined by the exterior portion 72. Similar to the internal microphone 20, the external microphone 70 is small (e.g., less than two mm in size). Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. Nemirovski ’379 at 10:8–13 The unique signals generated by the internal and external microphone 20, 70 can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 10′, the system 10′ can be adapted to cancel externally generated noise from the detections of the internal microphone 20. Nemirovski ’379 at 10:14–26 When the user speaks both the internal microphone 20 and the external microphone 70 detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle,

Claims	Prior Art
	machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 70 and the internal microphone 20 detect the noise. Accordingly, the communication device 14 (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 12 by the communication device 14. Nemirovski '379 at 10:27–47 In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or "*", represents a convolution operator): [Equation] $F_{ME}*S+F_{ME}*N=Y_E$ Eq. 2 By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[8a] A head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to determine an ambient sound level; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to determine an ambient sound level."

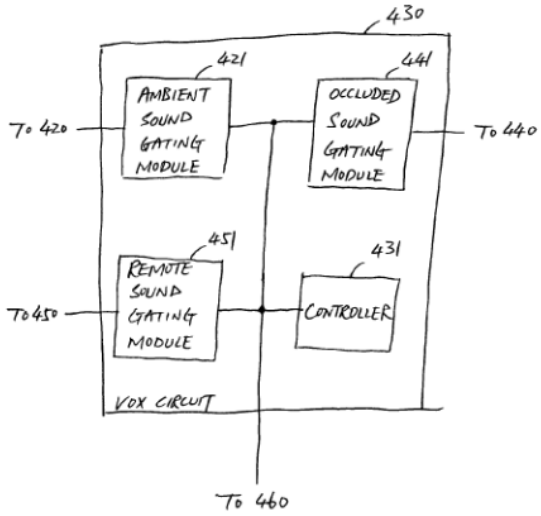
Claims	Prior Art
	Phonak – Perseo Brochure at Page 13 Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer. Phonak – Perseo Brochure at Page 23  Perseo® BTEs – reliable beauty The Perseo BTE family includes a full choice of sizes, colors and power levels. All feature an elegant, award-winning design with a refined, ergonomic shape. They have a patented housing construction with bonded plastic and silicon to effectively keep out debris and moisture. Environmental protection extends further with patented, disposable microphone covers to ensure consistent, reliable performance, in all conditions. With dual-microphone systems this is particularly important since electronic matching cannot successfully compensate for performance reduction as a result of debris and moisture. A comprehensive range of colors for the housing also caters for specific tastes and trends. Original size New small Perseo BTE 111/211

Claims	Prior Art
	Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user. Valente - Effect of type of noise and loudspeaker array on the performance at 399 **Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed**

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 6

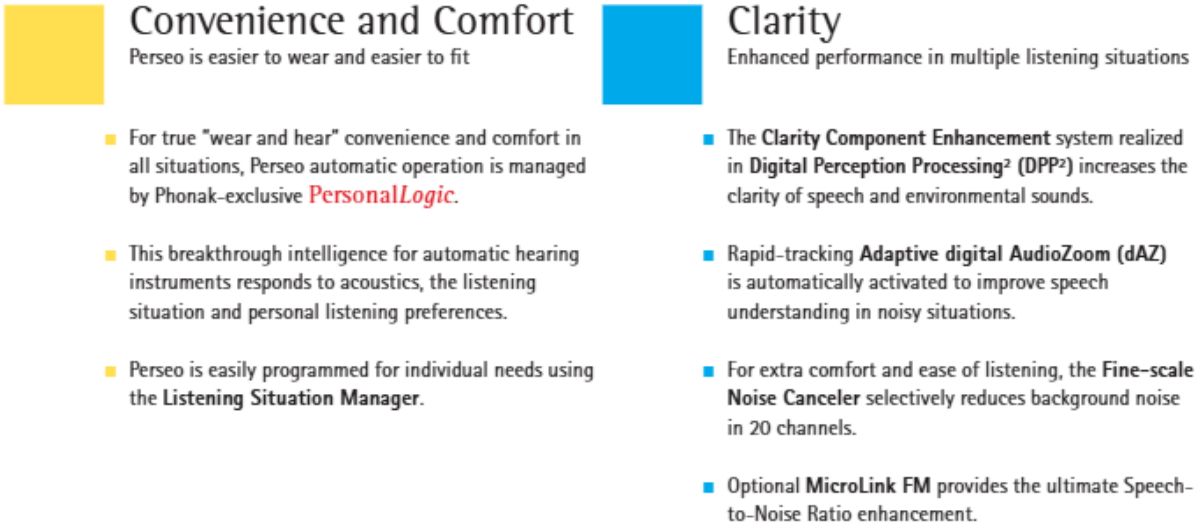


Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7  Adaptive digital AudioZoom (dAZ) Fine-scale Noise Canceler (FNC) **Active and adaptive separation of speech from noise** The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically. **Fully automatic, fully adaptive** With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR. **Twenty-band noise control** Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal. **Precise and selective** All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected. **Intelligent** The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved. 7 Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 DigiFocus' computer chips can remember different sound environments and adjust for them, depending on what listening situation the person is experiencing. Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone

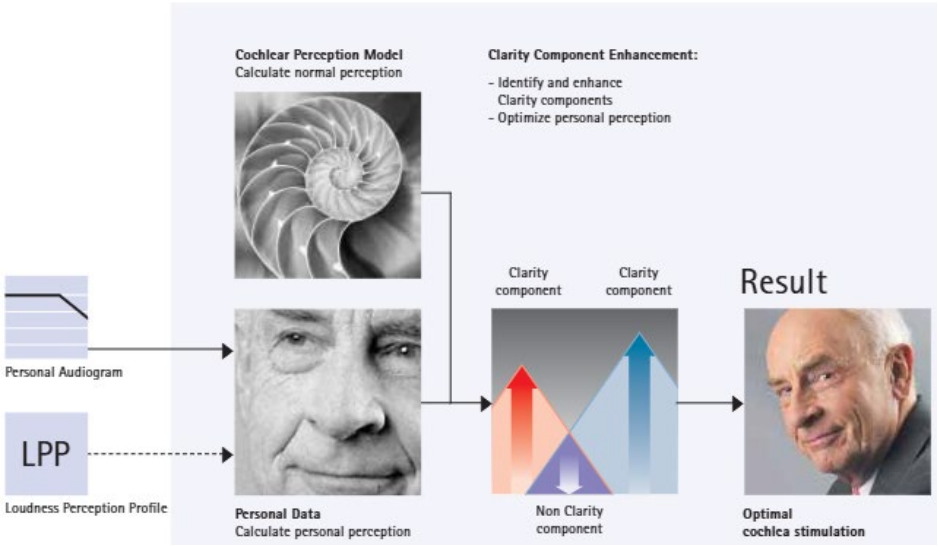
Claims	Prior Art
	banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Victorian '313 at FIG. 4B  FIG. 4B Victorian '313 at [0021] FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single ear-level device of FIG. 1. Victorian '313 at [0044] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an ear canal

Claims	Prior Art
	speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not "isolated" when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed

Claims	Prior Art
	for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110. Victorian '313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian '313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. A POSITA would have been motivated to combine the teachings of DHA systems and Victorian to meet the claim limitations with a reasonable expectation of success. A POSITA would have been motivated to combine the teachings of DHA systems and Nemirovski as described above with respect to claim 7. A POSITA would have further been motivated to combine the teachings of DHA systems with Victorian. DHA systems discloses an ear bud having a microphone that captures both ambient noise and a user's voice. The ear bud then produces a signal that is representative of ambient noise in the aided ear bud. A POSITA would recognize that it would be useful to employ the techniques disclosed in Victorian (e.g., to determine an ambient sound level) in order to determine how much the ambient noise signal should be reduced to produce

Claims	Prior Art
	the mixed signal. Moreover, the devices disclosed in DHA systems and Victorian share various structural and functional similarities that render the inventions readily combinable. For example, both DHA systems and Victorian disclose ear-worn devices for providing audio output to a hearing-impaired user. Both devices include similar components, such as microphones, speakers, wireless transceivers, and control circuitry. Accordingly, combining the teachings of DHA systems and Victorian would involve combining known components to yield predictable results. Furthermore, Nemirovski and Victorian are not incompatible. Both Nemirovski and Victorian disclose head-worn devices for supporting audio output. Nemirovski, FIG. 1; Victorian, FIG. 4A. The devices disclosed in Nemirovski and Victorian also share a number of familiar components, such as microphones, speakers, and transceiver units. Nemirovski, FIG. 1; Victorian, FIG. 4A. Accordingly, Nemirovski and Victorian are readily combinable. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[8b] modify audio output to the at least one speaker dependent at least in part on the ambient sound level.	DHA systems, alone or in view of Anderson, Jannard, Nemirovski, or Victorian, expressly or inherently disclose "modify audio output to the at least one speaker dependent at least in part on the ambient sound level." Phonak – Perseo Brochure at Page 7  Convenience and Comfort Perseo is easier to wear and easier to fit Clarity Enhanced performance in multiple listening situations For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. Perseo is easily programmed for individual needs using the **Listening Situation Manager**. The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 11 ■ With **dual-base automatic** the fitter has direct access to the two separate base-programs. Performance can be matched to each user's needs in quiet situations without disrupting performance in noisy situations and vice versa. The result is less fine-tuning effort than with single-base instruments. With **single-base automatic** time is wasted "juggling" a single set of fitting parameters while attempting to balance performance in multiple listening situations. ■ To further simplify fitting, PersonalLogic is pre-set according to typical needs and is easily re-programmed to individual preferences using the **Listening Situation Manager (LSM)**. Ensuring that an automatic hearing system responds correctly in noisy situations where there is no speech is an important consideration for safety and to eliminate feelings of isolation. The **LSM** provides this functionality with a single mouse click. Matching the system's sensitivity to noise with that of the client's, and controlling the auditory contrast between quiet and noisy settings is equally easy. Phonak – Perseo Brochure at Page 13 Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer.

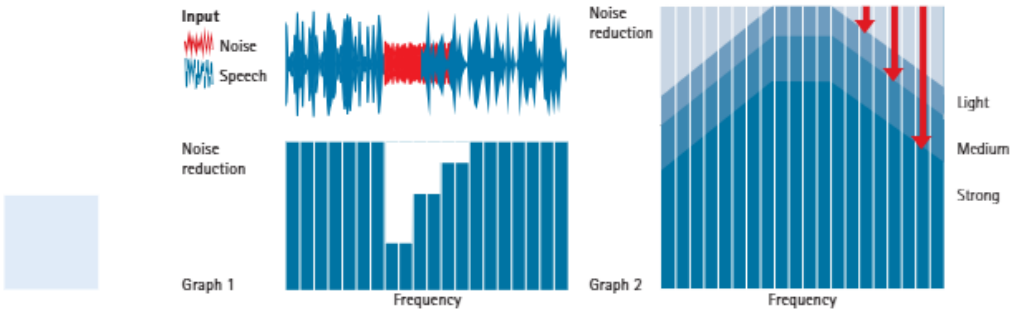
Claims	Prior Art
	Phonak – Perseo Brochure at Page 13 Digital Perception Processing² (DPP²)  Phonak – Perseo Brochure at p. 14 **Rapid-tracking dAZ** Phonak pioneered the development of directional microphone systems. Not surprisingly, Perseo **dAZ** with new soft switching brings even better performance. Perseo **dAZ** is automatically activated in appropriate situations and immediately reduces noise from beside and behind the wearer. The wearer experiences an immediate reduction of background noise and enhanced speech clarity. Rapid tracking ensures that even fast moving noise sources are controlled.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 14

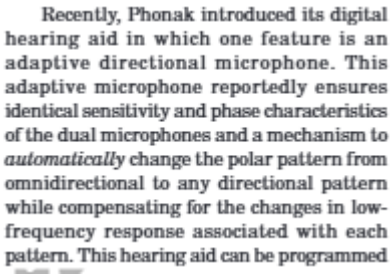
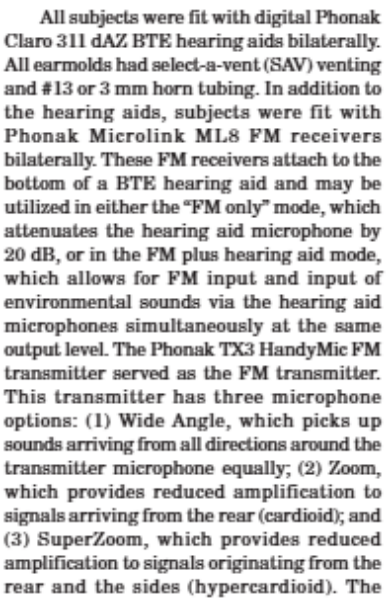
Enhanced comfort in all situations

Fine-scale Noise Canceler in 20 channels

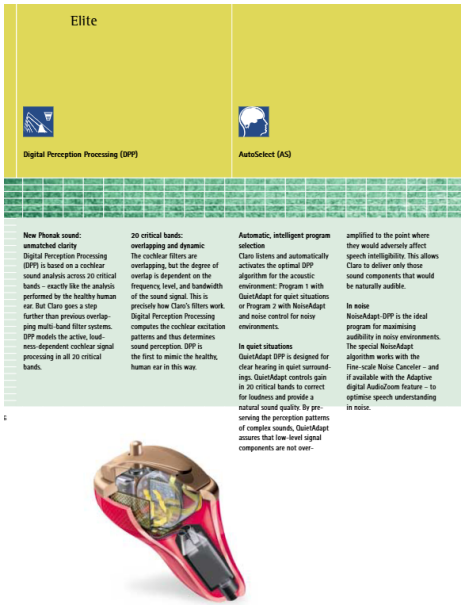
Finely scaled frequency resolution in 20 channels gives the **Fine-scale Noise Canceler (FNC)** the most precise background noise reduction capability. Noise reduction depends on the relative amounts of speech and noise in each channel (graph 1) and the importance of each channel for speech understanding. The **FNC** in Perseo ensures maximum comfort and ease of listening. Annoying background sounds are reduced for comfort while the important sounds of speech are preserved for stress-free listening. The maximum amount of noise reduction can be fitter-programmed to match the individual wearer's tastes and requirements. Thanks to the **dual-base automatic**, separate settings can be made for relatively quiet and noisy situations, ensuring optimal comfort in multiple situations (graph 2).



Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399  Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 431  All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 6



Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

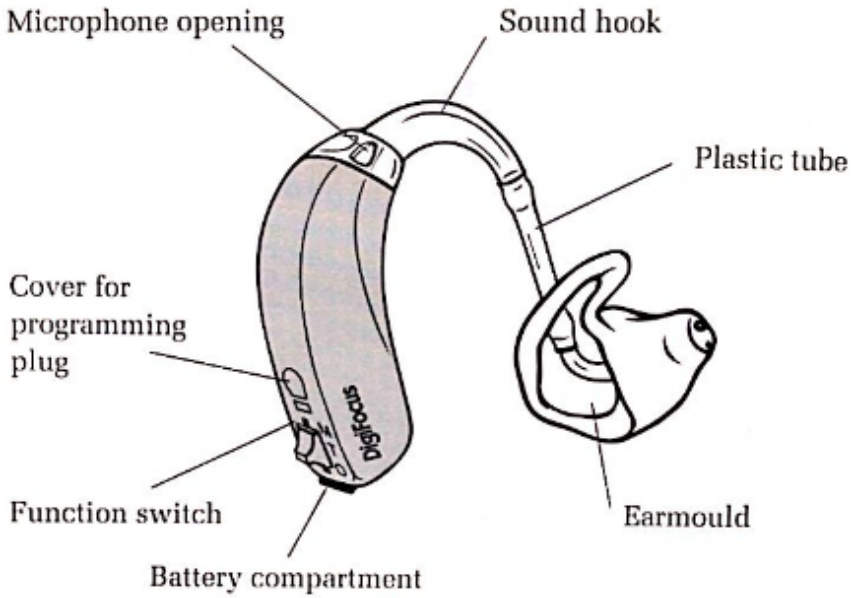
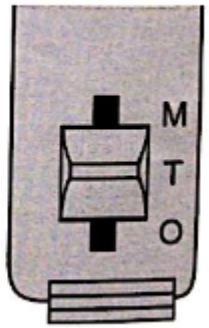
Fine-scale Noise Canceler (FNC)

Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

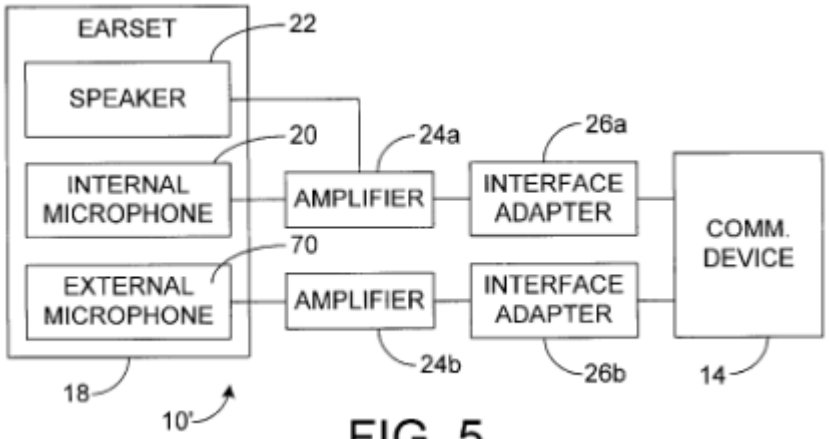
Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

7

Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Oticon – DigiFocus Compact User Instructional Brochure at Page 4 **Function switch** **M:** Hearing instrument switched **on** and microphone connected. **Normal use.** **T:** Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise. **O:** Hearing instrument switched **off.**  Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 "Today's amplifiers are not smart," said Peter Hahn, managing director of Oticon's U.S. headquarters in Somerset, N.J. "The digital processor can organize frequencies and manipulate speech signals to separate

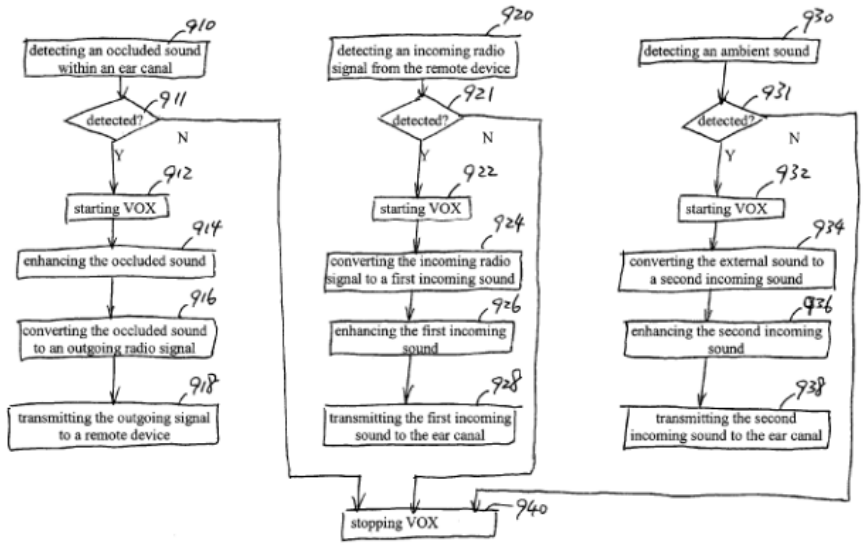
Claims	Prior Art
	them from background noise." Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 DigiFocus' computer chips can remember different sound environments and adjust for them, depending on what listening situation the person is experiencing. Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Oticon – News Release re Audiologists Test Breakthrough In Hearing Care (June 19, 2006) at Page 2 Audiologists at the Grunewald-Blitz Children's Clinic at The University of Arizona in Tucson are testing what promises to be a long-awaited breakthrough in hearing care -- a fully automatic compact hearing aid that digitally processes sound. It adjusts automatically to different noise levels. Oticon – News Release re Audiologists Test Breakthrough In Hearing Care (June 19, 2006) at Page 2 What makes the new digital hearing instrument a true breakthrough is its flexibility in processing sound, he added. "Think of the difference between a radio with only bass and treble controls and a fully-featured studio mixer, which adjusts sounds in many different frequency bands. Because of the miniature computer in DigiFocus, sound can be split into seven frequency bands, each of which can be adjusted in several ways. These bands are then processed through two channels to automatically adjust for loudness levels and speech." Anderson '783 at 23:65–24:14 Background noise and competing talker cancellation are achieved using any of a number of techniques and circuits well known in the art, e.g., variations of the Least Mean Squares (LMS) technique as implemented in the Motorola DSP56200 LMS filter integrated circuit, or the equivalent function implemented in the general-purpose RPU DSP 948. Note that for all binaural signal processing functions, data contained in replies from the left earpiece (i.e., audio data from the left earpiece microphone) is processed in the RPU DSP 948 by combining said data with other data contained in replies from the right

Claims	Prior Art
	earpiece (i.e., audio data from the right earpiece microphone). The processed audio data transmitted from the RPU to the left and right earpiece speakers via RPU interrogations is therefore a function of data received from both the left and right earpiece microphones via earpiece replies. Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes.

Claims	Prior Art
	Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone. Nemirovski ’379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from

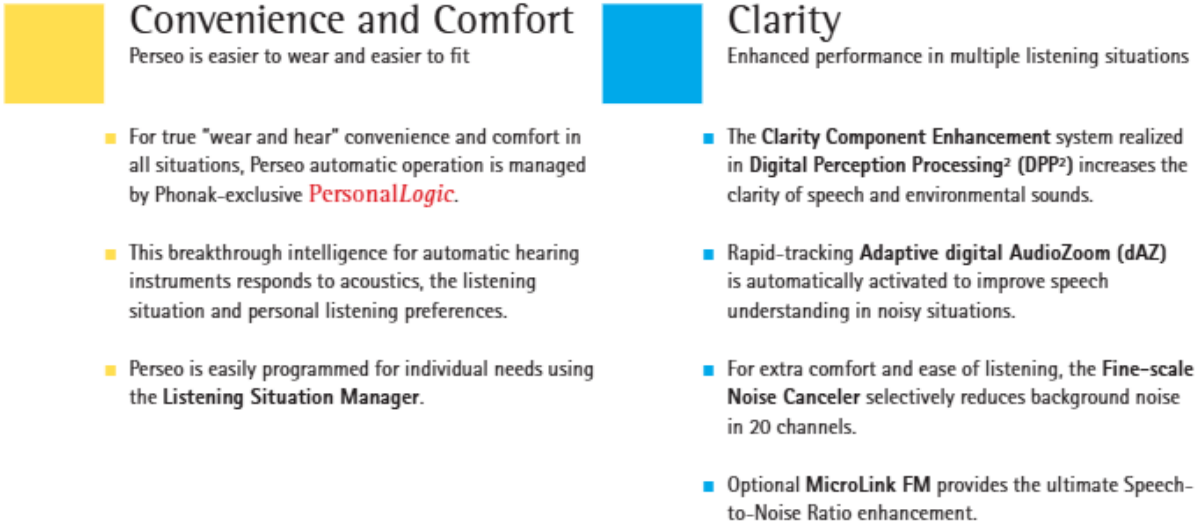
Claims	Prior Art
	the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. Nemirovski '379 at 9:33–39 The external microphone 70 is disposed on a exterior portion 72 of the housing 30 and is pointed away from the wearer of the earset 18 assembly. The external microphone 70 can be secured to the housing 30 with adhesive and can be disposed in a recess defined by the exterior portion 72. Similar to the internal microphone 20, the external microphone 70 is small (e.g., less than two mm in size). Nemirovski '379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. Nemirovski '379 at 10:8–13 The unique signals generated by the internal and external microphone 20, 70 can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 10′, the system 10′ can be adapted to cancel externally generated noise from the detections of the internal microphone 20. Nemirovski '379 at 10:14–26 When the user speaks both the internal microphone 20 and the external microphone 70 detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle,

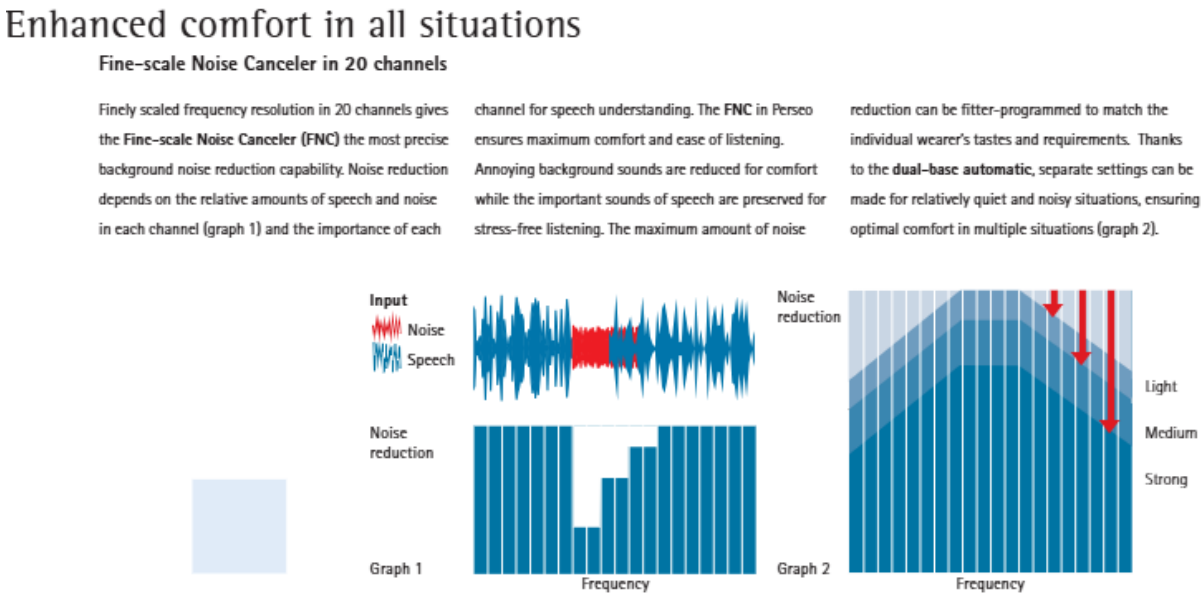
Claims	Prior Art
	machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 70 and the internal microphone 20 detect the noise. Accordingly, the communication device 14 (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 12 by the communication device 14. Nemirovski '379 at 10:27–47 In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or "*", represents a convolution operator): [Equation] $F_{ME}*S+F_{ME}*N=Y_E$ Eq. 2 By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N.

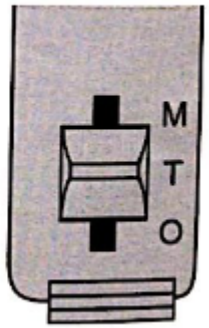
Claims	Prior Art
	Victorian ’313 at FIG. 9  FIG. 9 Victorian ’313 at [0026] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device. Victorian ’313 at [0044] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an ear canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not “isolated” when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals

Claims	Prior Art
	received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110. Victorian ’313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the

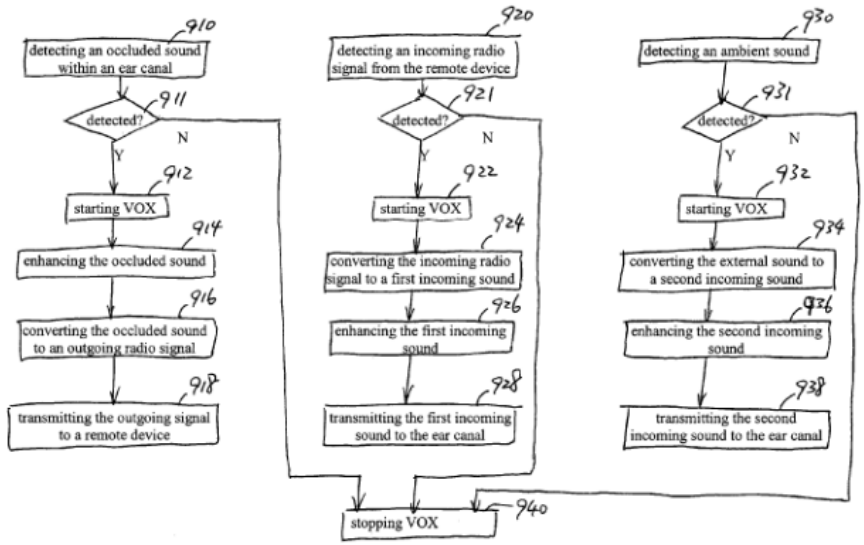
Claims	Prior Art
	ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian '313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. Victorian '313 at [0066] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, Nemirovski, or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 8a. To the extent that DHA systems, alone or in view of Anderson, Jannard, Nemirovski, or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[9a] A head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input." Phonak – Perseo Brochure at Page 7  **Convenience and Comfort** Perseo is easier to wear and easier to fit ■ For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. ■ This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. ■ Perseo is easily programmed for individual needs using the **Listening Situation Manager**. **Clarity** Enhanced performance in multiple listening situations ■ The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. ■ Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. ■ For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. ■ Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement.

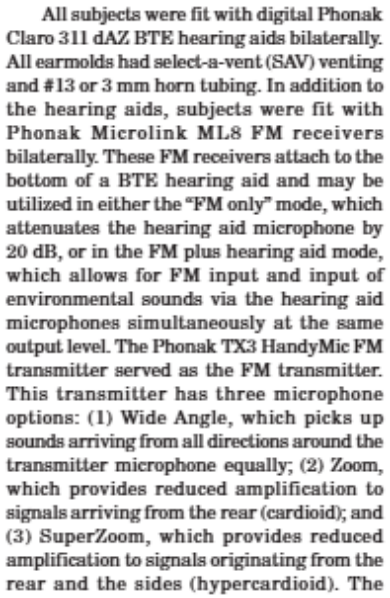
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	Phonak – Perseo Brochure at Page 14  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

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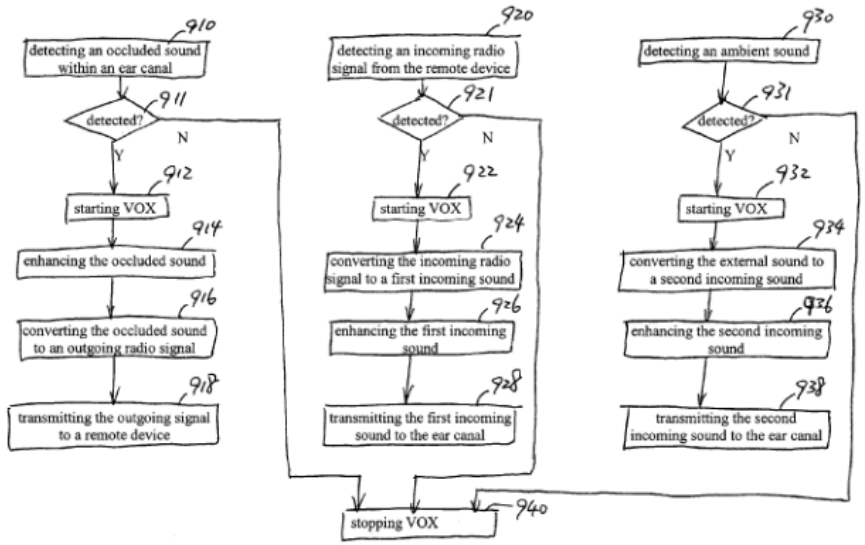
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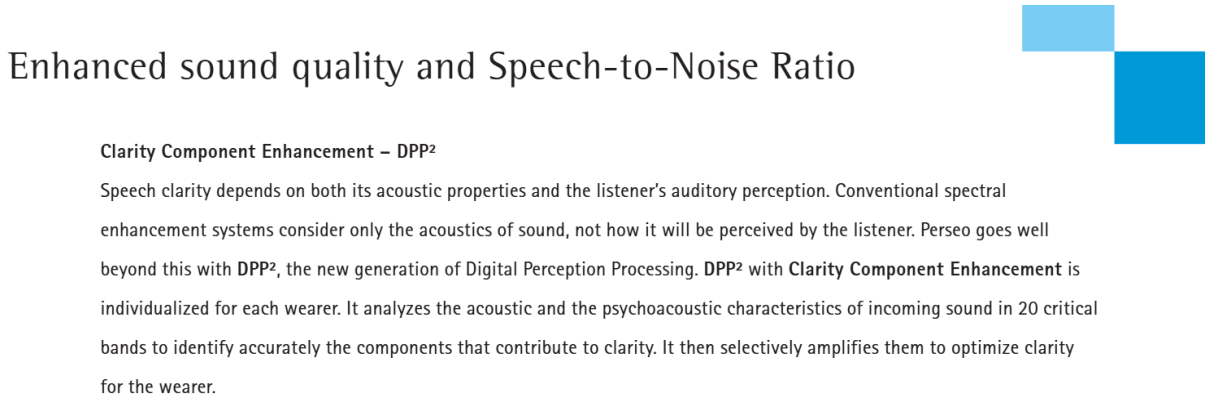
Claims	Prior Art
	enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Victorian '313 at FIG. 9  FIG. 9

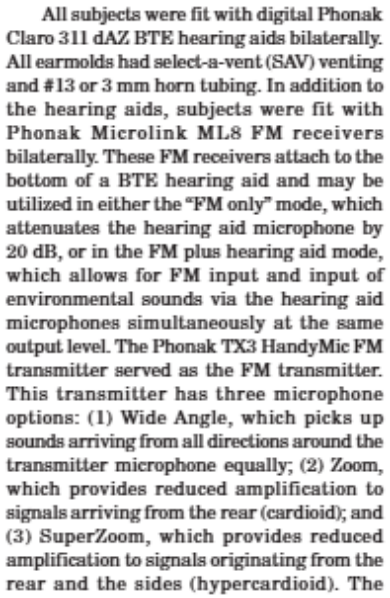
Claims	Prior Art
	Victorian ’313 at [0026] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device. Victorian ’313 at [0066] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 8a. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[9b] provide at least the ambient audio output signal to the at least one speaker.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose “provide at least the ambient audio output signal to the at least one speaker.”

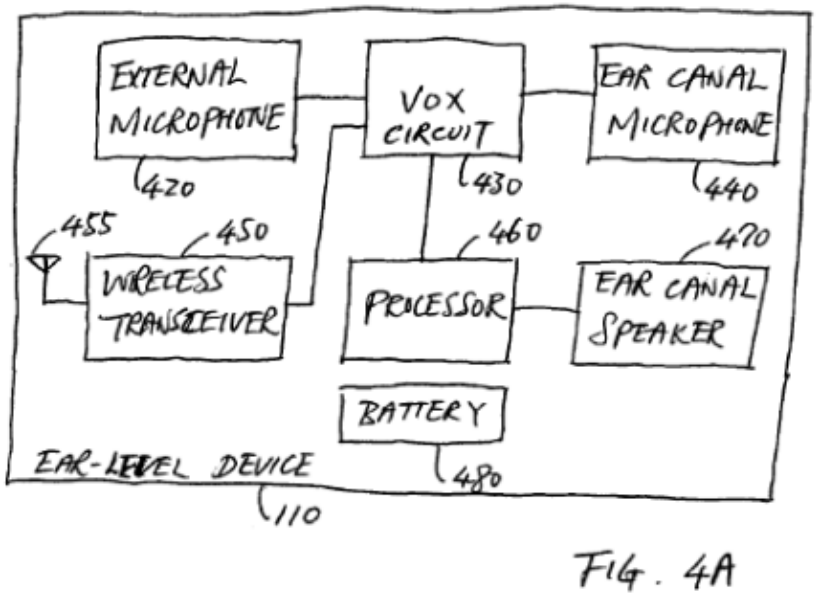
Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431  All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The Jannard '526 at 10:51–57 Further, some wearers of the audio device 10A′ may prefer to press the speakers 14A′, 16A′ into contact with the outer surfaces of their ears. For example, some users may desire to experience to loudest possible volume from the speakers 14A′, 16A′. Thus, by pressing the speakers 14A′, 16A′ against their ears, the perceived volume of the sound emitted from the speakers 14A′, 16A′ will be the greatest. Jannard '526 at 18:1–22 The microphone 75 can be disposed anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″. Optionally, the microphone 75 can be in the form of a bone conduction microphone. As such, the microphone 75 is disposed such that the when a user wears the audio device 10A′, the microphone 75 is in contact with the user's head 18. For example, but without limitation, the microphone can be positioned anywhere on the anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″ such that the microphone contacts the user's head. More preferably, the microphone 75 is positioned such that it contacts a portion of the user's head 18 near a bone, such that vibrations generated from the user's voice and traveling through the bone, are conducted to the microphone. In another alternative, the microphone 75 can be configured to be

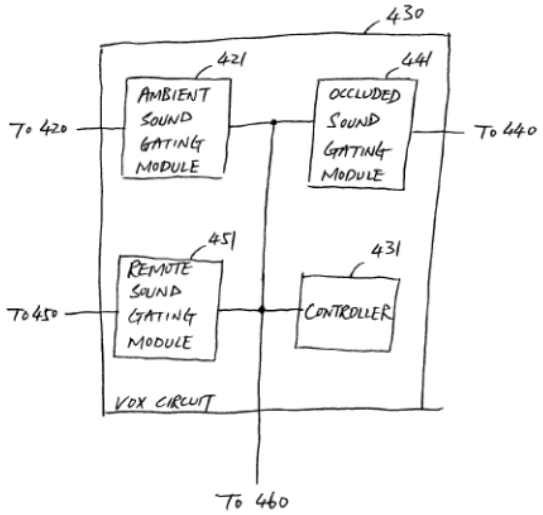
Claims	Prior Art
	inserted into the meatus 24 (FIG. 2) of the ear canal of the user. Thus, in this modification, the microphone 75 can be substituted for one of the speakers 14, 16. Alternatively, an ear-canal type bone conduction microphone can be combined with a speaker so as to provide two-way communication with the user through a single ear canal. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Jannard ’526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E.

Claims	Prior Art
	Victorian ’313 at FIG. 9  FIG. 9 Victorian ’313 at [0026] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device. Victorian ’313 at [0066] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 8a.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[10a] A head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input,	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 7, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[10b] wherein the ambient audio output signal is dependent on the ambient noise level.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the ambient audio output signal is dependent on the ambient noise level." Phonak – Perseo Brochure at Page 13 

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431  All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be

Claims	Prior Art
	supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Victorian '313 at FIG. 4A  FIG. 4A

Claims	Prior Art
	Victorian ’313 at FIG. 4B  FIG. 4B Victorian ’313 at [0020] FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device of FIG. 1. Victorian ’313 at [0021] FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single ear-level device of FIG. 1. Victorian ’313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote

Claims	Prior Art
	sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian '313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 8a. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[11a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a first microphone; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a first microphone." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[11b] a second microphone;	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski expressly or inherently disclose "a second microphone." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[11c] wherein the first microphone is configured to pick up at least audio input from at least the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the first microphone is configured to pick up at least audio input from at least the user." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[11d] wherein the second microphone is configured to pick up at least environmental audio input associated with an environment in the vicinity of the head-worn wearable electronic device, and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the second microphone is configured to pick up at least environmental audio input associated with an environment in the vicinity of the head-worn wearable electronic device." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to

Claims	Prior Art
	combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[11e] wherein the electronic circuitry is configured to process the audio input from at least the user picked-up via the first microphone to remove at least a portion of the environmental audio input associated with the environment picked-up via the second microphone.	DHA systems, alone or in view of Anderson, Jannard, Nemirovski, or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to process the audio input from at least the user picked-up via the first microphone to remove at least a portion of the environmental audio input associated with the environment picked-up via the second microphone." *See* above at claim [8b]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, Nemirovski, or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, Nemirovski, or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[12a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn wearable electronic device comprises at least a first microphone and a second microphone,	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn wearable electronic device comprises at least a first microphone and a second microphone."

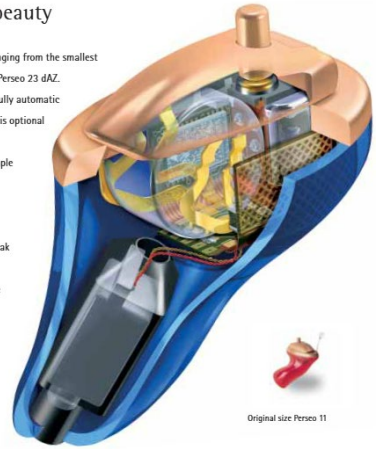
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.



Original size Perseo 11

IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 223

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

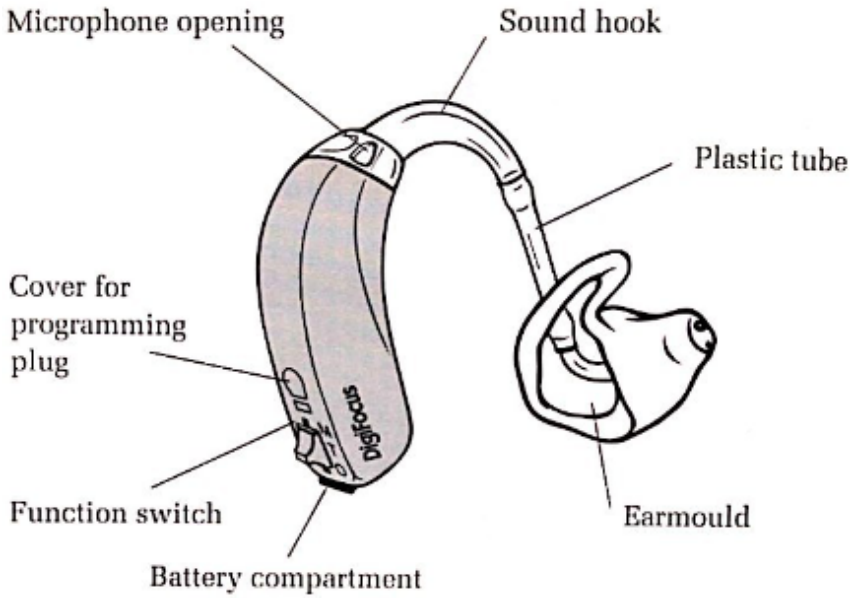
Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

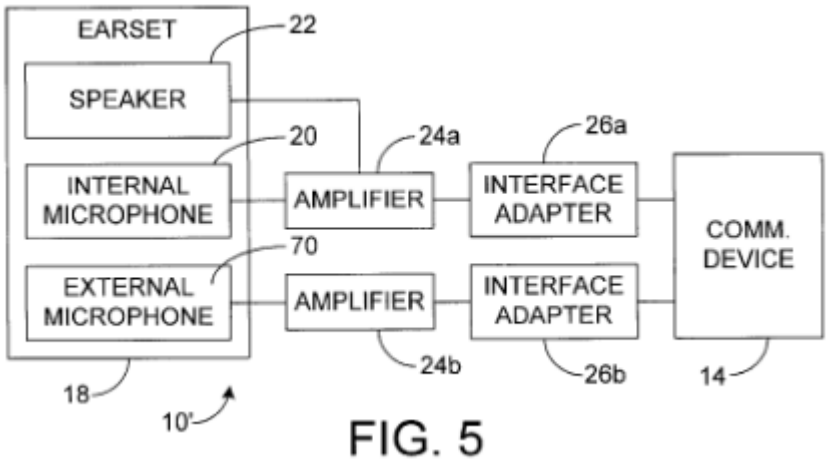
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

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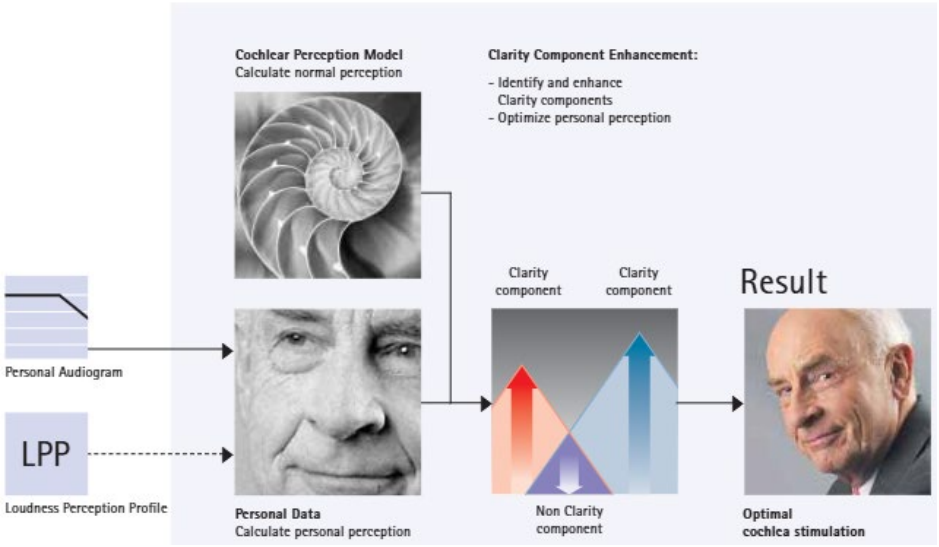
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Jannard ’526 at 2:26–34 In accordance with another embodiment of at least one of the inventions disclosed herein, an eyeglass frame comprises a support for supporting at least one lens in the path of a wearer's field of view, a first ear stem attached to the support, a second ear stem attached to the support, and at least one microphone supported by at least one of the support, first ear stem, and second ear stem. The microphone is advantageously arranged to face towards the head of a wearer of the eyeglass frame. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice

Claims	Prior Art
	recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski '379 at FIG. 5  FIG. 5 Nemirovski '379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski '379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely

Claims	Prior Art
	located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone. Nemirovski '379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Jannard or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[12b] wherein the first microphone is configured to pick up at least audio sound from at least the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the first microphone is configured to pick up at least audio sound from at least the user." *See* above at claim [7a].

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[12c] wherein the second microphone is configured to pick up at least ambient noise associated with an environment in the vicinity of the head-worn wearable electronic device, and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the second microphone is configured to pick up at least ambient noise associated with an environment in the vicinity of the head-worn wearable electronic device." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claim 7a. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[12d] wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient noise; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient noise." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 7a and 8a. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[12e] modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level.	DHA systems, alone or in view of Jannard, Nemirovski or Victorian, expressly or inherently disclose "modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level." Phonak – Perseo Brochure at Page 7 **Convenience and Comfort** Perseo is easier to wear and easier to fit ■ For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. ■ This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. ■ Perseo is easily programmed for individual needs using the **Listening Situation Manager**. **Clarity** Enhanced performance in multiple listening situations ■ The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. ■ Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. ■ For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. ■ Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement.

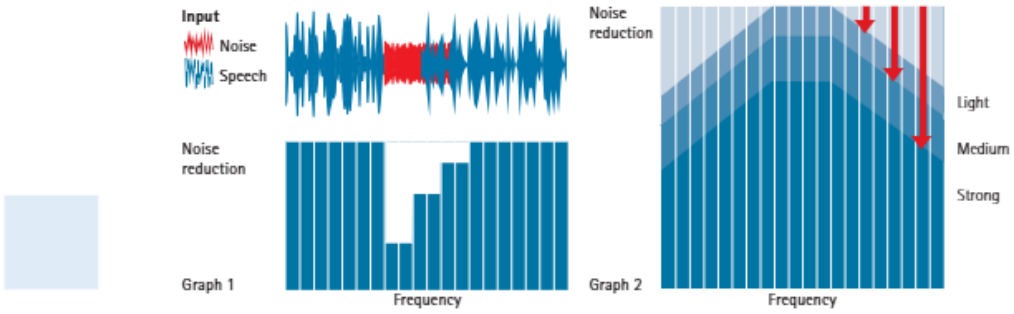
Claims	Prior Art
	Phonak – Perseo Brochure at Page 13 Digital Perception Processing² (DPP²) 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 14

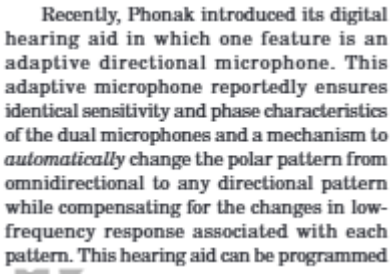
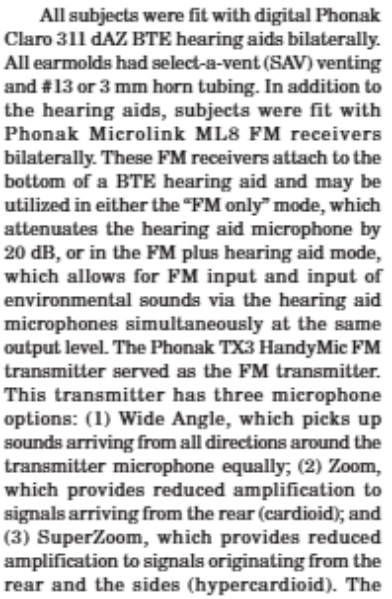
Enhanced comfort in all situations

Fine-scale Noise Canceler in 20 channels

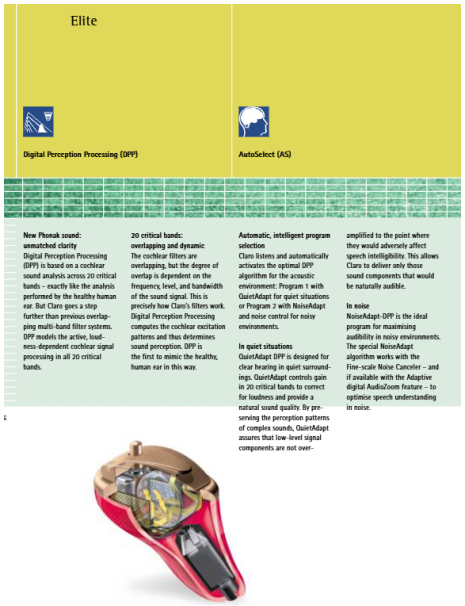
Finely scaled frequency resolution in 20 channels gives the **Fine-scale Noise Canceler (FNC)** the most precise background noise reduction capability. Noise reduction depends on the relative amounts of speech and noise in each channel (graph 1) and the importance of each channel for speech understanding. The **FNC** in Perseo ensures maximum comfort and ease of listening. Annoying background sounds are reduced for comfort while the important sounds of speech are preserved for stress-free listening. The maximum amount of noise reduction can be fitter-programmed to match the individual wearer's tastes and requirements. Thanks to the **dual-base automatic**, separate settings can be made for relatively quiet and noisy situations, ensuring optimal comfort in multiple situations (graph 2).



Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399  Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 431  All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 6



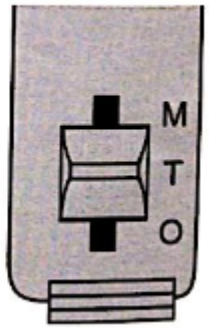
Oticon – DigiFocus Compact User Instructional Brochure at Page 4

Function switch

M: Hearing instrument switched **on** and microphone connected. **Normal use.**

T: Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise.

O: Hearing instrument switched **off.**



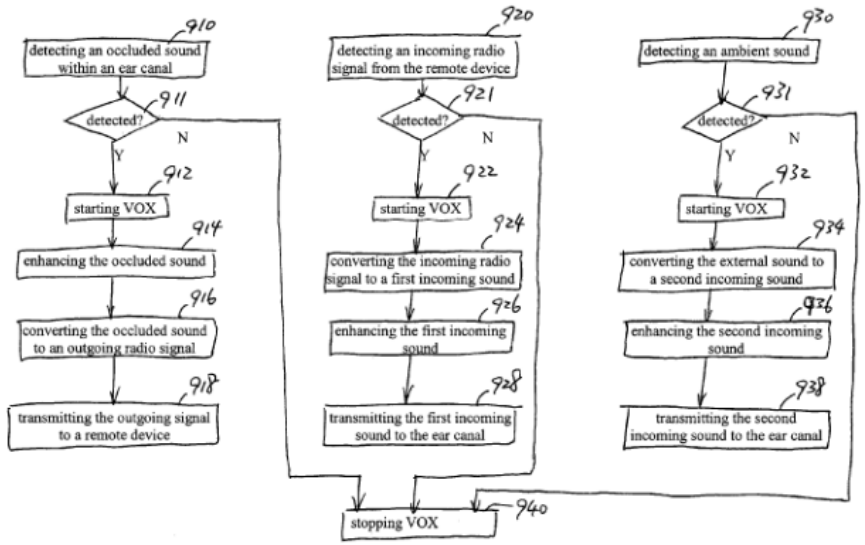
Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1

"Today's amplifiers are not smart," said Peter Hahn, managing director of Oticon's U.S. headquarters in Somerset, N.J. "The digital processor can organize frequencies and manipulate speech signals to separate

Claims	Prior Art
	them from background noise." Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 DigiFocus' computer chips can remember different sound environments and adjust for them, depending on what listening situation the person is experiencing. Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Oticon – News Release re Audiologists Test Breakthrough In Hearing Care (June 19, 2006) at Page 2 Audiologists at the Grunewald-Blitz Children's Clinic at The University of Arizona in Tucson are testing what promises to be a long-awaited breakthrough in hearing care -- a fully automatic compact hearing aid that digitally processes sound. It adjusts automatically to different noise levels. Oticon – News Release re Audiologists Test Breakthrough In Hearing Care (June 19, 2006) at Page 2 What makes the new digital hearing instrument a true breakthrough is its flexibility in processing sound, he added. "Think of the difference between a radio with only bass and treble controls and a fully-featured studio mixer, which adjusts sounds in many different frequency bands. Because of the miniature computer in DigiFocus, sound can be split into seven frequency bands, each of which can be adjusted in several ways. These bands are then processed through two channels to automatically adjust for loudness levels and speech." Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously

Claims	Prior Art
	enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. Nemirovski ’379 at 10:8–13 The unique signals generated by the internal and external microphone 20, 70 can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 10′, the system 10′ can be adapted to cancel externally generated noise from the detections of the internal microphone 20. Nemirovski ’379 at 10:14–26 When the user speaks both the internal microphone 20 and the external microphone 70 detects the speech.

Claims	Prior Art
	Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle, machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 70 and the internal microphone 20 detect the noise. Accordingly, the communication device 14 (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 12 by the communication device 14. Nemirovski '379 at 10:27–47 In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or "*", represents a convolution operator): [Equation] $F_{ME}*S+F_{ME}*N=Y_E$ Eq. 2 By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N.

Claims	Prior Art
	Victorian ’313 at FIG. 9  FIG. 9 Victorian ’313 at [0026] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device. Victorian ’313 at [0044] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an ear canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not “isolated” when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals

Claims	Prior Art
	received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110. Victorian ’313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the

Claims	Prior Art
	ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian '313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. Victorian '313 at [0066] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard, Nemirovski, or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claim 7a and 8a. To the extent that DHA systems, alone or in view of Jannard, Nemirovski, or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[13a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data,	DHA systems in view of Anderson, Jannard, or Abreu or Luo or Irvin expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data." Anderson '783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948. Jannard '526 at 26:21–27 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the transceiver 114 is adapted to employ a wide variety of technologies, including wireless communication such as RF, IR, ultrasonic, laser or optical, as well as wired and other communications technologies. In one embodiment, a body-LAN radio is employed. Other embodiments can employ a flexible-circuit design. Jannard '526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance. Abreu '960 at 6:28–38 Head-mounted structures are structures mounted on the head or neck for positioning sensors on the end of the tunnel and include head bands with accessories that are adjacent to the tunnel, visors, helmets, headphone, structures wrapping around the ear and the like. For the purpose of this description TempAlert is referred herein as a system that measures temperature in the BTT area and has means to report the measured value and that can incorporate alarm means that are activated when certain levels are reached. Support structures yet include any article that has sensing means in which said sensing means are positioned at the end of the tunnel.

Claims	Prior Art
	Abreu ’960 at 10:49–55 The apparatus of the invention include a temperature sensor located in the medial canthal pad. For temperature measurement the sensing system is located on a skin area that includes the medial canthal corner of the eye and upper eyelid. The sensor in the medial canthal pad is preferably positioned on the skin adjacent to the medial canthal area (medial corner of the eye). Abreu ’960 at 32:29–45 In accordance with this exemplary embodiment for temperature measurement, the thermal energy emitted by the BTT is sensed by the temperature sensor 102 such as a miniature thermistor which produces a signal representing the thermal energy sensed. The signal is then converted to digital information and processed by processor 106 using standard processing for determining the temperature. Luo ’446 at [0036] The invention generally relies on shining light having a particular wavelength on human skin as well as the reflectance properties of human skin. The surface reflection of the human skin occurs at the surface of the epidermis and is approximately independent of the lighting wavelength and independent of human race. In the infrared (IR) range the wavelengths that can be used are greater than 800 nano-meters and the surface reflectance of human skin is more or less constant and close to 50%. The same hold true for orange and red light in the visible light range, namely wavelengths between approximately 600 and 800 nano-meters can be used, although the surface reflectance for darker skin is reduced for shorter wavelengths. Surface reflectance for dark skin is approximately 25% at 700 nano-meters. In the embodiments shown herein, the energy of choice in the sensor unit 18 is preferably infrared (IR) energy. However, long wavelength visible light energy can also be used as discussed below. Irvin ’332 at 6:3–33 FIG. 4C illustrates the power control circuit 210 coupled with a contact switch functioning as the sensor 220. The contact switch includes a first and second contact connected to the power control circuit 210, and connecting member 222 actuated when the user places the wireless headset 100 on their body. When actuated, the connecting member 222 places the first and second contacts of sensor 220 in electrical connection. Ideally, the sensor 220, when configured as a contact sensor, is placed in the earpiece 130, in the retaining member 140, or in the retaining pad 150 shown in FIG. 1B. Preferably, when sensor 220 is configured as a contact-based proximity sensor, the contacting element of sensor 220 is sealed in a manner that minimizes contaminant ingress. In some embodiments, the contact sensor may be actuated by the user resiliently expanding the retaining member 140 or other element of the wireless headset 100 for

Claims	Prior Art
	mounting upon their body. FIG. 4D illustrates the power control circuit 210 coupled with a proximity sensor 220 not based on mechanical actuation. The proximity sensor 220 may be an active or passive IR sensor, may be a resistive or other type of thermal sensor, or may be an acoustic sensor. As a proximity sensor, sensor 220 provides indication that a user has placed the wireless headset 100 upon their body. As such, any of the foregoing or other presence sensing techniques may be used to implement sensor 220. Preferably, when configured as a proximity sensor, sensor 220 requires minimal power and provides a sensor output signal based on proximity to the user's body. Depending upon the sensing technology chosen, the control circuit 210 may be required to periodically stimulate sensor 220 to generate a detection response. A POSITA would have been motivated to combine the teachings of the DHA system with Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. In addition, a POSITA would have been motivated to combine the teachings of the DHA system with Anderson, Jannard, or Abreu or Luo or Irvin to meet the claim limitations with a reasonable expectation of success. For example, the DHA system, Anderson, Jannard, and Abreu or Luo or Irvin each disclose head-worn devices. The Phonak system discloses that its device can include a sensor (a microphone), so it would have been obvious to the POSITA to incorporate other types of sensors into the device as well. Given that Anderson, Jannard, and Abreu discloses a head-worn device with various sensors for measuring different types of data, the POSITA would have looked to Abreu or Luo or Irvin for different types of sensors that could be incorporated into the Phonak system. Moreover, a POSITA would have understood that sending a signal if the earpiece is no longer being worn by a user would necessarily include processing the data from the sensor to make that determination. Further, a POSITA would have understood that the sensor would fail to capture signals in a proper manner if the earbud were not being worn. Accordingly, Abreu's system could determine whether the Phonak system's earbud are being worn based on at least the sensor data. Further, a POSITA would have been motivated to modify the DHA system to include this feature to conserve battery use, as if the earbud is not being worn, the device can switch to a low power mode and minimize power usage. Moreover, the devices of the DHA system and Abreu or Luo or Irvin share various structural and functional similarities that render the inventions readily combinable. Accordingly, combining the teachings of the DHA system with Anderson, Jannard, or Abreu or Luo or Irvin would involve combining known components to yield predictable results. To the extent that the DHA system, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, does not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[13b] wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and	DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, expressly or inherently disclose "wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector." Anderson '783 at 12:47–13:22 Remote processor unit interrogator and reply processor: The RPU, shown in FIG. 9, may use any of a wide variety of components well known in the art. The DSP 948 may contain several integrated circuits, e.g., the Motorola DSP56L002 single-chip digital signal processor, a Motorola DSP56200 cascadable-adaptive digital filter chip and supporting memory devices well known in the art. Miniature components such as dielectric resonance oscillators are available for the microwave oscillator 908, and the DPSK replies can be decoded using surface acoustic wave (SAW) or glass bulk acoustic wave (BAW) delay lines 936 in a standard delay-and-multiply demodulator circuit 934. An alternative demodulator that achieves superior noise immunity can be implemented using a phase-locked loop in a squaring loop configuration for carrier recovery instead of delay-and-multiply circuitry 934. Decoding of a DPSK waveform using such coherent detection techniques is known in the art as differentially encoded coherent phase shift keying, or DECPSK. Amplifiers using low noise pseudomorphic high electron mobility transistors (e.g., the HP ATF-36077) may be used as needed (e.g., between the circulator 906 and routing switch 922), and such amplifiers are not individually identified in FIG. 9. All switches shown in FIG. 9 are RF switches that can be implemented using, e.g., PIN diodes. Switch control connections from the DSP 948 to the RF switches are shown in FIG. 9 for clarity, and all switches remain set as shown until changed as noted. The system shown in FIG. 9 demodulates a received reply when the RF reply travels through a selected antenna 900, antenna selection switch 902, circulator 906, routing switch 922 (set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 976), bandpass filter 928 and IF mixer 930. The IF mixer 930 downconverts the reply by mixing it with a small amount of RF power from the 5760 MHz oscillator 908 via switch 910 (which is set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 972) and power splitter 920. The resulting DPSK reply has a center frequency of 60 MHz, a bandwidth of 100 MHz, and is passed through the bandpass filter 932 to the DPSK demodulator 934 which, in this example, consists of a 50 nsec delay line 936 and multiplier 938. The resulting bits are stored in the reply data register 940 for processing by the DSP 948. Anderson '783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid

Claims	Prior Art
	system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948. Anderson '783 at 24:26–32 Sensor measurements can be reported to the user by a voice synthesized in the RPU DSP 948 using any of a variety of synthesis techniques well known in the art, including playback of pre-recorded digitized speech. The voice signal produced within the RPU DSP 948 is then added to the processed audio signals already being sent from the RPU to the earpiece. Abreu '960 at 6:28–38 Head-mounted structures are structures mounted on the head or neck for positioning sensors on the end of the tunnel and include head bands with accessories that are adjacent to the tunnel, visors, helmets, headphone, structures wrapping around the ear and the like. For the purpose of this description TempAlert is referred herein as a system that measures temperature in the BTT area and has means to report the measured value and that can incorporate alarm means that are activated when certain levels are reached. Support structures yet include any article that has sensing means in which said sensing means are positioned at the end of the tunnel. Abreu '960 at 10:49–55 The apparatus of the invention include a temperature sensor located in the medial canthal pad. For temperature measurement the sensing system is located on a skin area that includes the medial canthal corner of the eye and upper eyelid. The sensor in the medial canthal pad is preferably positioned on the skin adjacent to the medial canthal area (medial corner of the eye). Abreu '960 at 15:57–60 The eyeglasses can be incorporated with a detecting system to send a signal if the eyeglasses were removed or if the temperature sensor is not capturing signals in a proper manner. Abreu '960 at 18:56–59 The present invention preferably includes all functions in a miniature semiconductor chip, which as an

Claims	Prior Art
	integrated circuit, incorporates sensor, processing and transmitting units and control circuits. Abreu ’960 at 32:29–45 In accordance with this exemplary embodiment for temperature measurement, the thermal energy emitted by the BTT is sensed by the temperature sensor 102 such as a miniature thermistor which produces a signal representing the thermal energy sensed. The signal is then converted to digital information and processed by processor 106 using standard processing for determining the temperature. Luo ’446 at [0036] The invention generally relies on shining light having a particular wavelength on human skin as well as the reflectance properties of human skin. The surface reflection of the human skin occurs at the surface of the epidermis and is approximately independent of the lighting wavelength and independent of human race. In the infrared (IR) range the wavelengths that can be used are greater than 800 nano-meters and the surface reflectance of human skin is more or less constant and close to 50%. The same hold true for orange and red light in the visible light range, namely wavelengths between approximately 600 and 800 nano-meters can be used, although the surface reflectance for darker skin is reduced for shorter wavelengths. Surface reflectance for dark skin is approximately 25% at 700 nano-meters. In the embodiments shown herein, the energy of choice in the sensor unit 18 is preferably infrared (IR) energy. However, long wavelength visible light energy can also be used as discussed below. Irvin ’332 at 6:3–33 FIG. 4C illustrates the power control circuit 210 coupled with a contact switch functioning as the sensor 220. The contact switch includes a first and second contact connected to the power control circuit 210, and connecting member 222 actuated when the user places the wireless headset 100 on their body. When actuated, the connecting member 222 places the first and second contacts of sensor 220 in electrical connection. Ideally, the sensor 220, when configured as a contact sensor, is placed in the earpiece 130, in the retaining member 140, or in the retaining pad 150 shown in FIG. 1B. Preferably, when sensor 220 is configured as a contact-based proximity sensor, the contacting element of sensor 220 is sealed in a manner that minimizes contaminant ingress. In some embodiments, the contact sensor may be actuated by the user resiliently expanding the retaining member 140 or other element of the wireless headset 100 for mounting upon their body. FIG. 4D illustrates the power control circuit 210 coupled with a proximity sensor 220 not based on mechanical actuation. The proximity sensor 220 may be an active or passive IR sensor, may be a resistive or other type of thermal sensor, or may be an acoustic sensor. As a proximity sensor, sensor 220 provides indication that a user has placed the wireless headset 100 upon their body. As such, any of the foregoing

Claims	Prior Art
	or other presence sensing techniques may be used to implement sensor 220. Preferably, when configured as a proximity sensor, sensor 220 requires minimal power and provides a sensor output signal based on proximity to the user's body. Depending upon the sensing technology chosen, the control circuit 210 may be required to periodically stimulate sensor 220 to generate a detection response. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson or Abreu or Luo or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claim 13a. To the extent that DHA, alone or in view of Anderson or Abreu or Luo or Irvin, systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[13c] wherein the electronic circuitry is configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data.	DHA systems, alone or in view of Abreu or Luo or Irvin, expressly or inherently disclose "wherein the electronic circuitry is configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data." Abreu '960 at 15:57–60 The eyeglasses can be incorporated with a detecting system to send a signal if the eyeglasses were removed or if the temperature sensor is not capturing signals in a proper manner. Abreu '960 at 18:56–59 The present invention preferably includes all functions in a miniature semiconductor chip, which as an integrated circuit, incorporates sensor, processing and transmitting units and control circuits. Abreu '960 at 44:61–67 The present invention includes means for directing preferably near-infrared energy into the surface of the skin at the end of the BTE tunnel, means for analyzing and converting the reflectance or back scattered spectrun into the concentration of the substance measured and support means for positioning the light source and detector means adjacent to the surface of the skin at the BTE tunnel site. Luo '446 at [0034] In general, the location sensor module 18 and the hearing aid module 24 communicate via a bi-directional

Claims	Prior Art
	information signal 42. The hearing aid module 24 polls the location sensor module 18 via the bi-directional signal 42, on preferably a periodic basis, to determine whether the hearing aid 10 is in the ear of the hearing aid user. In this case, the bi-directional signal 42 acts as a polling signal. In response to polling done via the bi-directional information signal 42, the location sensor module 18 probes the outer environment of the hearing aid 10 and returns location information via the bi-directional location signal 42. In this case, the bi-directional signal 42 acts as a location information signal. If the location information indicates that the hearing aid 10 is in the ear of the hearing aid user (i.e. the in-the-ear case), the hearing aid 10 operates in full-function mode. If the location information indicates that the hearing aid 10 is not in the ear of the hearing aid user (i.e. the out-of-the-ear case), the hearing aid 10 operates in sleep mode. In sleep mode, the hearing aid 10 can essentially be considered to be off. Luo '446 at [0035] There are several general scenarios for the location of the hearing aid. In the first scenario, the hearing aid 10 is not in the ear of the user and the hearing aid 10 and is in sleep mode. In this case, the location sensor module 18 is polled and the location information indicates that the hearing aid 10 is not in the ear of the hearing aid user and the hearing aid 10 continues to operate in sleep mode. In the second scenario, the hearing aid 10 has just been inserted into the ear of the hearing aid user and the hearing aid 10 was previously in sleep mode. In this case, the location sensor module 18 is polled and the location information indicates that the hearing aid 10 is in the ear of the hearing aid user. The hearing aid 10 then moves into full-function mode. In the third scenario, the hearing aid 10 is in full-function mode and is being taken out of the ear of the hearing aid user. The location sensor module 18 is polled and the location information indicates that the hearing aid 10 is no longer in the ear of the hearing aid user. The hearing aid 10 then moves into sleep mode. Luo '446 at [0036] The invention generally relies on shining light having a particular wavelength on human skin as well as the reflectance properties of human skin. The surface reflection of the human skin occurs at the surface of the epidermis and is approximately independent of the lighting wavelength and independent of human race. In the infrared (IR) range the wavelengths that can be used are greater than 800 nano-meters and the surface reflectance of human skin is more or less constant and close to 50%. The same hold true for orange and red light in the visible light range, namely wavelengths between approximately 600 and 800 nano-meters can be used, although the surface reflectance for darker skin is reduced for shorter wavelengths. Surface reflectance for dark skin is approximately 25% at 700 nano-meters. In the embodiments shown herein, the energy of choice in the sensor unit 18 is preferably infrared (IR) energy. However, long wavelength visible light energy can also be used as discussed below.

Claims	Prior Art
	Irvin ’332 at 2:20–34 The wireless headset serves as a communication accessory to a base unit, such as a cellular telephone. When not in use, the wireless headset reverts to an inactive state and draws essentially no power. A power control circuit imparts automatic enabling capability to the wireless headset. Preferably, the power control circuit employs a sensor responsive to a natural stimulus associated with a person preparing to use the wireless headset. The wireless headset may sense motion, physical orientation, or user proximity. Sensing techniques include but are not limited to micro-electro-mechanical-systems (MEMS), mercury switches, attitude switches, and IR or acoustic proximity sensors. In some embodiments, the wireless headset transitions from the inactive state to an active state in response to a user picking it up or mounting the wireless headset upon their head. Irvin ’332 at 6:3–33 FIG. 4C illustrates the power control circuit 210 coupled with a contact switch functioning as the sensor 220. The contact switch includes a first and second contact connected to the power control circuit 210, and connecting member 222 actuated when the user places the wireless headset 100 on their body. When actuated, the connecting member 222 places the first and second contacts of sensor 220 in electrical connection. Ideally, the sensor 220, when configured as a contact sensor, is placed in the earpiece 130, in the retaining member 140, or in the retaining pad 150 shown in FIG. 1B. Preferably, when sensor 220 is configured as a contact-based proximity sensor, the contacting element of sensor 220 is sealed in a manner that minimizes contaminant ingress. In some embodiments, the contact sensor may be actuated by the user resiliently expanding the retaining member 140 or other element of the wireless headset 100 for mounting upon their body. FIG. 4D illustrates the power control circuit 210 coupled with a proximity sensor 220 not based on mechanical actuation. The proximity sensor 220 may be an active or passive IR sensor, may be a resistive or other type of thermal sensor, or may be an acoustic sensor. As a proximity sensor, sensor 220 provides indication that a user has placed the wireless headset 100 upon their body. As such, any of the foregoing or other presence sensing techniques may be used to implement sensor 220. Preferably, when configured as a proximity sensor, sensor 220 requires minimal power and provides a sensor output signal based on proximity to the user's body. Depending upon the sensing technology chosen, the control circuit 210 may be required to periodically stimulate sensor 220 to generate a detection response. Irvin ’332 at 8:11–22 FIGS. 6 and 7 illustrate operation of the wireless headset 100. FIG. 6 is a state diagram for one exemplary embodiment of the present invention. When the wireless headset 100 is not in use, it automatically enters an inactive state (610). As explained, this inactive state may constitute disconnecting the communications circuit 230 from the power source 240, or may simply represent placing the communications circuit 230 in

Claims	Prior Art
	a non-functional, low-power condition. Absent user stimulus, such as motion of the wireless headset 100 or placement of the wireless headset 100 upon the user's body, the wireless headset 100 remains in the inactive state (610). A POSITA would have been motivated to combine the teachings of DHA systems and Abreu or Luo or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claim 13a. To the extent that DHA systems, alone or in view of Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[14a] A head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus includes a force sensor configured to receive a user input based at least on sensing force applied to the force sensor, and	DHA systems in view of Abreu or Irvin expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 1, wherein the head-worn personal electronic apparatus includes a force sensor configured to receive a user input based at least on sensing force applied to the force sensor." Abreu '960 at 15:60–65 By way of illustration, but not of limitation, pressuse sensing means can be incorporated in the end of the temples to detect if the sunglasses are being worn, and an abrupt drop in the pressure signal indicates glasses were removed or misplacement of the sensor can also generate an identifiable signal. A POSITA would understand the "pressure sensing means" to be a "force sensor," since measuring pressure necessarily requires measuring force. Irvin '332 at 6:34–48 FIG. 4E illustrates the power control circuit 210 coupled to an alternate type of attitude sensor 220. In this embodiment, the attitude sensor 220 is a mechanical sensor that includes a contacting member 224 responsive to a physical orientation of the sensor 220. In one or more physical orientations, the contacting member 224 does not electrically connect the first and second contacts of attitude sensor 220. In one or more other physical orientations, the contacting member places the first and second contacts of attitude sensor 220 in electrical connection. The power control circuit 210 is configured to be responsive to this electrical connection. Contacting member 224 may be resiliently biased away from one or both contacts to ensure an open electrical connection between contacts when the headset is not oriented in a position indicative of use.

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of DHA systems and Abreu or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claim 13a. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[14b] wherein the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device.	DHA systems in view of Abreu or Irvin expressly or inherently disclose "wherein the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device." Abreu '960 at 15:60–65 By way of illustration, but not of limitation, pressuse sensing means can be incorporated in the end of the temples to detect if the sunglasses are being worn, and an abrupt drop in the pressure signal indicates glasses were removed or misplacement of the sensor can also generate an identifiable signal. Irvin '332 at 2:20–34 The wireless headset serves as a communication accessory to a base unit, such as a cellular telephone. When not in use, the wireless headset reverts to an inactive state and draws essentially no power. A power control circuit imparts automatic enabling capability to the wireless headset. Preferably, the power control circuit employs a sensor responsive to a natural stimulus associated with a person preparing to use the wireless headset. The wireless headset may sense motion, physical orientation, or user proximity. Sensing techniques include but are not limited to micro-electro-mechanical-systems (MEMS), mercury switches, attitude switches, and IR or acoustic proximity sensors. In some embodiments, the wireless headset transitions from the inactive state to an active state in response to a user picking it up or mounting the wireless headset upon their head. Irvin '332 at 6:34–48 FIG. 4E illustrates the power control circuit 210 coupled to an alternate type of attitude sensor 220. In this embodiment, the attitude sensor 220 is a mechanical sensor that includes a contacting member 224 responsive to a physical orientation of the sensor 220. In one or more physical orientations, the contacting member 224 does not electrically connect the first and second contacts of attitude sensor 220. In one or more other physical orientations, the contacting member places the first and second contacts of attitude sensor 220 in electrical connection. The power control circuit 210 is configured to be responsive to this electrical connection. Contacting member 224 may be resiliently biased away from one or both contacts to

Claims	Prior Art
	ensure an open electrical connection between contacts when the headset is not oriented in a position indicative of use. Irvin '332 at 8:11–22 FIGS. 6 and 7 illustrate operation of the wireless headset 100. FIG. 6 is a state diagram for one exemplary embodiment of the present invention. When the wireless headset 100 is not in use, it automatically enters an inactive state (610). As explained, this inactive state may constitute disconnecting the communications circuit 230 from the power source 240, or may simply represent placing the communications circuit 230 in a non-functional, low-power condition. Absent user stimulus, such as motion of the wireless headset 100 or placement of the wireless headset 100 upon the user's body, the wireless headset 100 remains in the inactive state (610). A POSITA would have been motivated to combine the teachings of DHA systems and Abreu or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claim 13a. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[15a] A head-worn personal electronic apparatus as recited in claim 14, wherein the head-worn personal electronic apparatus comprises: a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 14, wherein the head-worn personal electronic apparatus comprises: a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claims [1pre] and [7a]. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

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[15b] a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input associated with an environment around the head-worn personal electronic apparatus,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input associated with an environment around the head-worn personal electronic apparatus." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Nemirovski, with a reasonable expectation of success, for the reasons described above with respect to claims [1pre] and [7a]. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[15c] wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone.	DHA systems, alone or in view of Jannard or Nemirovski '379 or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone." *See* above at claim [11e]. A POSITA would have been motivated to combine the teachings of DHA systems and Jannard or Nemirovski '379 or Victorian, with a reasonable expectation of success, for the reasons described above with respect to claims [1pre], [7], and [8]. To the extent that DHA systems, alone or in view of Jannard or Nemirovski '379 or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[16a] A head-worn personal electronic apparatus as recited in claim 15, wherein the head-worn personal electronic apparatus comprises: a radiation detector	DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 15, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data." *See* above at claim [13a].

Claims	Prior Art
configured to capture radiation data,	A POSITA would have been motivated to combine the teachings of DHA systems and Anderson, Jannard, or Abreu or Luo or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claims [1pre] and [13]. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[16b] wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data.	DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, expressly or inherently disclose "wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data." *See* above at claims [13b] and [13c]. A POSITA would have been motivated to combine the teachings of DHA systems and Anderson or Abreu or Luo or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claims [1pre] and [13]. To the extent that DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[17pre] A head-worn personal electronic apparatus operable to at least output customized audio output, comprising:	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus operable to at least output customized audio output." *See* above at claim 1[pre]. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[17a] at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user;	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user." *See* above at claim [1a].

Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[17b] a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear; and	DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Radar, expressly or inherently disclose "a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear." Phonak – Perseo Brochure at Page 5 The Science of PersonalLogic™ Phonak has applied knowledge from new research and past experience to add a new dimension of individualized logic to automatic hearing systems – PersonalLogic. Combining new and proven features, PersonalLogic optimizes the functioning, operation and fitting of hearing systems for each individual user. Thanks to PersonalLogic, Perseo takes customization to a level beyond "custom-fitted" related to individual hearing loss. The new digital system from Phonak is the first hearing system with customized automatic operation, as opposed to operation based on factory set, unchangeable rules. Now, Perseo gives hearing professionals control over automatic operation, allowing them to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 15

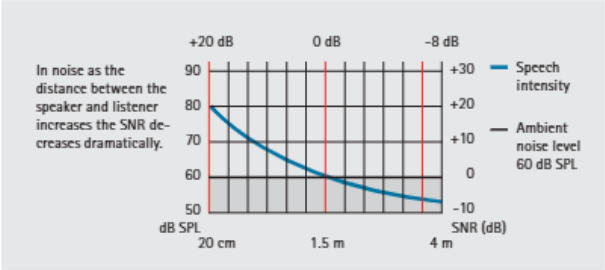
The perfectly integrated communication system

FM MicroLink

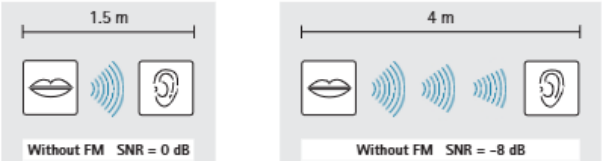
FM technology has allowed thousands of children to participate in standard education settings and to be actively involved in social activities. Phonak pioneered this development and the integration of FM into hearing instruments to produce communication systems that are effective in even the most demanding situations. Since the innovation of miniaturized Phonak **MicroLink** systems, FM has become an important solution also for adults with a high communication need in work and leisure environments.

All Perseo BTEs have been designed to incorporate MicroLink with its fully integrated appearance and operation. This integrated system is unique to Phonak.

Speech-to-Noise Ratio (SNR) as a function of distance between speaker and listener



Speech-to-Noise Ratio without FM is distance dependent

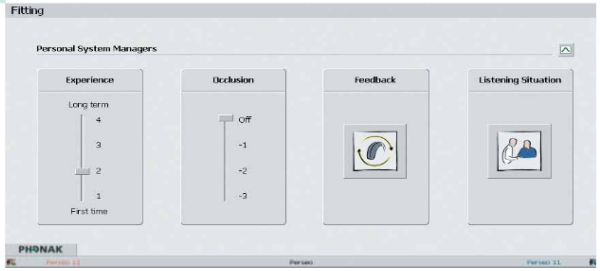


IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 258

Claims	Prior Art
	Phonak – Perseo Brochure at Page 18

The art of the personal touch – Personal System Managers

PFG 8.1 provides a set of **Personal System Managers** to take care of the technical details behind common fitting issues, leaving the fitter free to take care of the wearer. This approach ensures that fitting is uncomplicated. It is always clear which controls should be used and how to use them.



The **Experience Manager** adjusts tonality and loudness to match each client's experience with hearing instruments. An effective and convenient control to set and adjust the client's hearing system as they become accustomed to their new world of sound.

The **Occlusion Manager** eliminates the feeling of blocked ears experienced by some wearers.

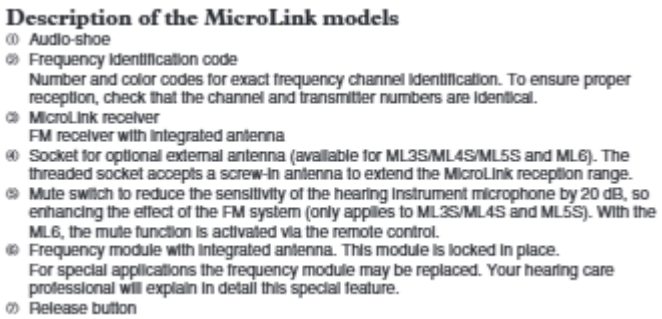
The dynamic **Feedback Manager** automatically identifies any bands at risk from feedback. Processing strategies are modified in these bands to prevent feedback while also increasing audibility for soft sounds. The result is easier listening without feedback.

The **Listening Situation Manager** allows the fitter to program *PersonalLogic* according to the wearer's preference for when to trigger a change of base programs, and the desired degree of sound contrast between the base-programs. Additional personalization is available, if required, by directly programming either of the base programs.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 20 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 28  Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak – Microlink User Guide at Page 2 

Claims	Prior Art
	Phonak – Microlink User Guide at Page 6 **The direct-input technique** Using the audio cable supplied with the system, the HandyMic can be connected directly to any sound system, including a TV, radio, walkman, movie projector, amplifier, cassette-player etc.  Phonak – Microlink User Guide at Page 25 **Two versions of MicroLink MLx** MicroLink MLx is available in two versions which are differentiated by the position of the plugs. The correct version for your needs is defined by the appropriate Audio-shoe.  MLx with plugs oriented lengthways MLx with plugs oriented transversely **Important:** Place the hearing instrument with audio-shoe and MicroLink MLx in front of you as shown in the sketch. The green dots must be visible at the top and bottom of the MicroLink MLx unit. If this is not the case please contact your hearing care professional. Correct MLx version fitted to hearing instrument  Wrong MLx version fitted to hearing instrument 25

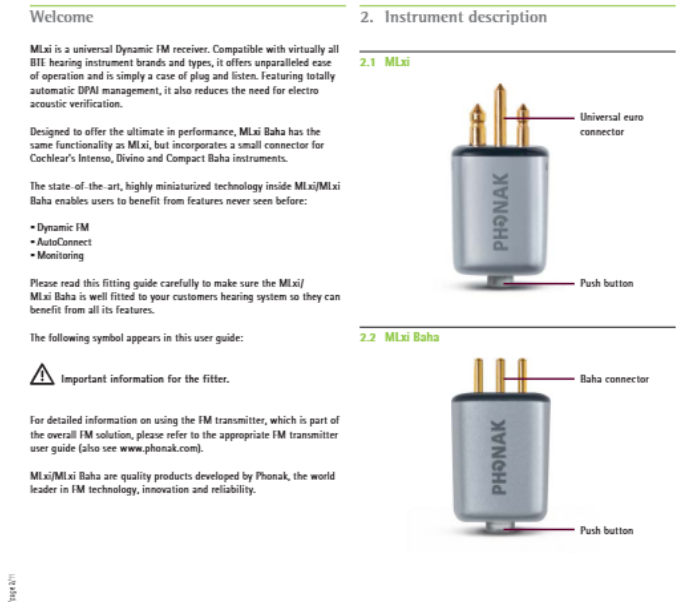
Claims	Prior Art
	Phonak – Microlink User Guide at Page 28  **Description of the MicroLink models** ① Audio-shoe ② Frequency identification code Number and color codes for exact frequency channel identification. To ensure proper reception, check that the channel and transmitter numbers are identical. ③ MicroLink receiver FM receiver with integrated antenna ④ Socket for optional external antenna (available for ML3S/ML4S/ML5S and ML6). The threaded socket accepts a screw-in antenna to extend the MicroLink reception range. ⑤ Mute switch to reduce the sensitivity of the hearing instrument microphone by 20 dB, so enhancing the effect of the FM system (only applies to ML3S/ML4S and ML5S). With the ML6, the mute function is activated via the remote control. ⑥ Frequency module with integrated antenna. This module is locked in place. For special applications the frequency module may be replaced. Your hearing care professional will explain in detail this special feature. ⑦ Release button Phonak – Microlink User Guide at p. 52 The HandyMic and MicroVox TX2 are used by hearing impaired people. To operate the HandyMic and MicroVox TX2, hearing instruments with FM receivers (e.g., MicroLink) are required. The user benefit of FM is an improved signal-to-noise ratio which leads to an optimized speech understanding even in very difficult listening situations. Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results.

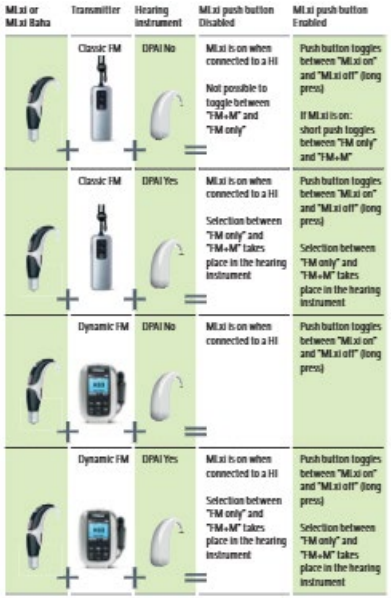
Claims	Prior Art
	Phonak – Online Interview re Perseo (April 7, 2003) at Page 9 **Liz Brassine/Phonak:** It is a digital microphone on the CIC. There are several advantages of a digital microphone over analog mics. First, increased dynamic range is provided. Additionally, with a digital mic, the analog-to-digital converter is housed within the microphone itself. This eliminates the wire between the AD-converter and the microphone. The wire is the source for ultra-high frequency interference, such as Bluetooth. Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink SX is the first communication device to incorporate • Advanced FM features • Hearing Instrument remote control • A Bluetooth link for cell phone use Phonak - SmartLink SX - FM Systems Manual at p. 4 Smart connections to all audio sources Any external audio input or an external microphone signal can be transmitted wirelessly via SmartLink to the hearing instrument, simply by attaching to the smart interface. Phonak - SmartLink SX - FM Systems Manual at p. 5 Smart FM transmission The proven Phonak FM technology clearly transmits all incoming auditory signals from SmartLink to the receiver attached to the user's hearing instrument.

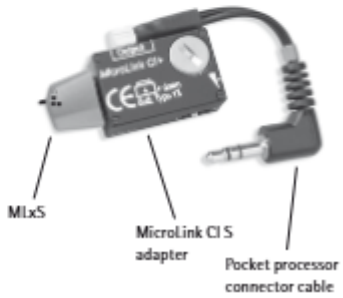
Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 10 **Smart links with Bluetooth** With the inclusion of Bluetooth technology in SmartLink, Phonak now enables hearing instrument wearers to enjoy unrestricted use of all Bluetooth compatible cellular phones. One of the major complaints of hearing impaired people is now history – talking on the telephone is even easier for them than for people with normal hearing. Simply synchronizing the Bluetooth telephone and SmartLink creates the communication path. As long as the cell phone is within a 10-meter range of SmartLink, all incoming calls are automatically directed via SmartLink to the hearing instrument. The cell phone is only needed for dialing outgoing calls. Whenever the cell phone is switched on, the Bluetooth function remains active for incoming calls, regardless of whether SmartLink is switched on or off. This guarantees that telephone calls are not missed, even when the FM function is not required. When using an external audio input to listen to music, incoming telephone calls have priority when the cell phone is switched on. The audio signal is automatically decreased so that the user hears the ringing tone clearly. Phonak - SmartLink SX - FM Systems Manual at p. 10wire  0

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 11 Smart digital synthesizer technology Digital frequency synthesizer technology, only available with Phonak FM communication systems, enables online frequency synchronization and scanning of the frequencies used. Once a frequency is selected on the SmartLink, it automatically sends a signal to the FM receiver that adjusts the frequency of the receiver accordingly. This automatic synchronization process occurs every time the SmartLink is switched on. In cases where more than one FM transmitter is in use, at lectures or presentations for example, the SmartLink has an exclusive scan function. This seeks out the frequency being used by the other transmitter and sets the receiver frequency accordingly. Phonak - SmartLink SX - FM Systems Manual at p. 14 

Claims	Prior Art
	Phonak - SmartLink SX User Guide at p. 6 This unique combination of an FM transmitter, mobile phone link and hearing instrument remote control is the gateway to a completely new hearing experience. Phonak - SmartLink SX User Guide at p. 32 

Claims	Prior Art
	Phonak – Fitting Guide for MLxi and MLxi Baha at Page 2  Welcome MLxi is a universal Dynamic FM receiver. Compatible with virtually all BTE hearing instrument brands and types, it offers unparalleled ease of operation and is simply a case of plug and listen. Featuring totally automatic DPAI management, it also reduces the need for electro acoustic verification. Designed to offer the ultimate in performance, MLxi Baha has the same functionality as MLxi, but incorporates a small connector for Cochlear's Intenso, Divino and Compact Baha instruments. The state-of-the-art, highly miniaturized technology inside MLxi/MLxi Baha enables users to benefit from features never seen before: • Dynamic FM • AutoConnect • Monitoring Please read this fitting guide carefully to make sure the MLxi/ MLxi Baha is well fitted to your customers hearing system so they can benefit from all its features. The following symbol appears in this user guide: Important information for the fitter. For detailed information on using the FM transmitter, which is part of the overall FM solution, please refer to the appropriate FM transmitter user guide (also see www.phonak.com). MLxi/MLxi Baha are quality products developed by Phonak, the world leader in FM technology, innovation and reliability. 2. Instrument description 2.1 MLxi Universal euro connector PHONAK Push button 2.2 MLxi Baha Baha connector PHONAK Push button Phonak – Fitting Guide for MLxi and MLxi Baha at Page 4  4.2 Advanced fitting of MLxi using FM SuccessWare The extended mode of FM SuccessWare enables you to fine-tune some parameters of the MLxi according to the needs of your customers. PHONAK

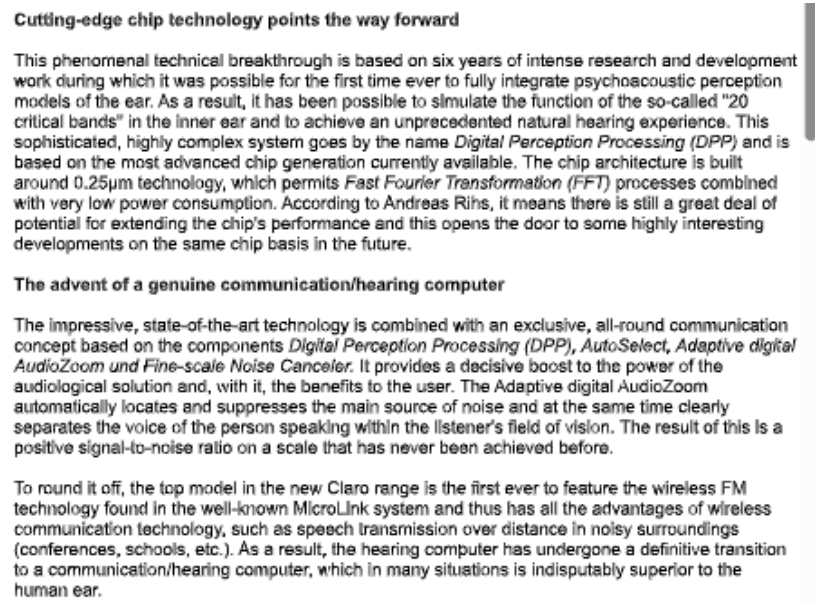
Claims	Prior Art
	Phonak – Fitting Guide for MLxi and MLxi Baha at Page 5 However, MLxi can also be used with the following Classic FM transmitters: Campus SX, EasyLink, ZoomLink, SmartLink and Classic FM inspiro. For these combinations, there is no Adaptive FM Advantage feature, which automatically adjusts the FM Advantage in noisy environments, and no Voice Activity Detection, which automatically reduces the FM level when no voice is present at the transmitter microphone. In those cases the user needs to have the chance to select between "Off", "FM only" and "FM+M", as with a Classic FM receiver. For DPAI Yes hearing instruments this selection takes place by toggling between different HI programs such as "no FM", "FM only" and "FM+M". For DPAI No hearing instruments this selection takes place on the receiver. Phonak – Fitting Guide for MLxi and MLxi Baha at Page 6 

Claims	Prior Art
	Phonak MicroLink MLxS Product Information at Page 1 The MicroLink MLxS is the first miniaturized FM receiver with a digital frequency synthesizer for full multi-frequency flexibility. The receiver features fully automatic, non-contact frequency synchronization with either a WallPilot or the Campus S. The MicroLink MLxS is compatible with all Phonak FM transmitters and the WallPilot, and is recommended for use with this equipment. **MLxS with BTE instrument**  **MLxS with adapter for CI pocket processor**  Phonak – Media Release re Product Offerings in F2003/2004 at Page 2 The key breakthrough technology in Perseo is Personal*Logic*, the first ever automatic control system that is fitter programmable according to the tastes and preferences of each individual wearer. Automatic performance has now a new level of intelligence beyond merely adapting to acoustic changes in the environment. Perseo is the first hearing system with customized automatic operation, as opposed to operation based on factory settings. It allows hearing professionals to provide clients with a hearing system "custom-tailored" to their personal hearing requirements and tastes.

Claims	Prior Art
	Phonak – Media Release re Product Offerings in F2003/2004 at Page 3 **Phonak Communications introduces the new MicroLink receivers with frequency synthesizer** Phonak Communications will introduce its new multi frequency MLx S and ML8 S receivers in the next few months. MLx S and ML8 S are the counterpart of Campus S transmitters, already introduced early 2002. Together, they comprise the new synthesized FM system generation, a system that transmits, without wires, speech to the hearing instrument in difficult listening situations. The key technology of the new system is its easy frequency adaptation. Given the extreme design constraints (low voltage, low power, miniature dimensions), the development of this technology represents an unprecedented breakthrough in wireless communication technology. Phonak – History of Various Hearing Aid Manufacturers at Page 5 **MLxS – The breakthrough in miniaturized multi-frequency FM** The digital frequency synthesizer is the heart of the revolutionary multi-frequency MLxS. This technology breakthrough, the result of years of meticulous development work, offers totally new communication possibilities. **2002** CEO Peter Pfluger leaves Phonak. New CEO Dr. Valentin Chapero Rueda joins the company on October 1, 2002. Phonak – History of Various Hearing Aid Manufacturers at Page 6 **Claro dAZ FM** Claro dAZ FM is the world's first digital hearing computer with design integrated FM receiver. The two most advanced systems for signal processing and FM signal reception now form a perfect combination of function and design. The result is optimized speech understanding in all adverse conditions involving distance, reverberation and noise.

Claims	Prior Art
	Phonak – History of Various Hearing Aid Manufacturers at Page 6 **MicroLink MLx – The open platform for Phonak and other brands** The first self-contained receiver with a standard interface enabling connection to all Phonak instruments and most other behind-the-ear hearing instruments. Phonak – The Hearing Review Article re Adults' Use of FM Systems at Page 2 An Example of an FM System for Adults The FM system used in this study was the MicroLink system by Phonak. This system has a receiver that is placed in an audio shoe or "boot," and the boot is snapped onto the hearing instrument over the battery door (Figure 1). Each manufacturer has its own boot, and the FM receiver (called MLx) is attached with three prongs that snap into the three holes of the boot. A switch on the bottom of the receiver offers control with the options "off," "FM-only," and "FM plus hearing aid."  *Figure 1. The MicroLink FM system snaps onto the hearing instrument over the battery door. Many manufacturers have their own boot. A switch on the bottom of the receiver allows "off," "FM only," and "FM plus hearing aid" modes.* The receiver is called the "universal receiver" because it can be used by different hearing instruments from different manufacturers. It can also be switched to a different hearing instrument from another manufacturer with a boot for that device. Therefore, if a patient is using "Hearing Aid A" with a MicroLink system, and the patient changes to "Hearing Aid B" from a different manufacturer, the MicroLink system can be used with the new aid. In this study, five different manufacturers' products were used, demonstrating the versatility of the system. (Author's Note: Many programmable or digital BTEs produced during the past six years are MicroLink compatible. Analog BTEs are usually compatible if they have direct audio input (DAI) capability.)

Claims	Prior Art
	Phonak – The Hearing Review Article re Adults' Use of FM Systems at Page 4 **Study Method** The study's 28 participants were all experienced users with behind-the-ear (BTE) hearing instruments. Of the 28 participants in the study, one patient was wearing analog aids, eight wore programmable, and 19 wore digital hearing aids. BTE devices used by the participants included: one Phonak analog power device; four Phonak 332 power AudioZoom; two Phonak 232 AudioZoom; two Phonak E-4s; three Siemens Prismas; three Siemens Signias; five Widex Senso's; four Resound BZ5s; one Oticon Digifocus; and three Phonak Claros. Twenty-two of the participants were binaural hearing instrument users and six were monaural hearing instrument users. Twenty participants were wearing devices with directional microphones. The majority of these patients utilized high-end technology devices with optimal SNR capabilities. They were already using exceptional devices that improved their hearing ability, but all were willing to try an FM system if there was the potential of improving their current benefit from the aids. Phonak – The Hearing Review Article re Adults' Use of FM Systems at Page 7 • A patient who uses 332 Power AudioZooms with 8% and 20% discrimination had not been able to use the phone for the past few years. After using the TX4, he has been able to communicate on the phone and he feels that his understanding on the phone is now about 95%. • A patient with a severe-to-profound unilateral hearing loss with 28% discrimination has been struggling to use the phone. He tried the TX4 and this allowed him to hear well on the phone for the first time in many years.

Claims	Prior Art
	Phonak Claro – World Premiere at Page 1 

Cutting-edge chip technology points the way forward

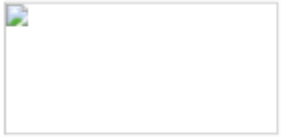
This phenomenal technical breakthrough is based on six years of intense research and development work during which it was possible for the first time ever to fully integrate psychoacoustic perception models of the ear. As a result, it has been possible to simulate the function of the so-called "20 critical bands" in the inner ear and to achieve an unprecedented natural hearing experience. This sophisticated, highly complex system goes by the name *Digital Perception Processing (DPP)* and is based on the most advanced chip generation currently available. The chip architecture is built around 0.25µm technology, which permits *Fast Fourier Transformation (FFT)* processes combined with very low power consumption. According to Andreas Rihs, it means there is still a great deal of potential for extending the chip's performance and this opens the door to some highly interesting developments on the same chip basis in the future.

The advent of a genuine communication/hearing computer

The impressive, state-of-the-art technology is combined with an exclusive, all-round communication concept based on the components *Digital Perception Processing (DPP), AutoSelect, Adaptive digital AudioZoom und Fine-scale Noise Canceler.* It provides a decisive boost to the power of the audiological solution and, with it, the benefits to the user. The Adaptive digital AudioZoom automatically locates and suppresses the main source of noise and at the same time clearly separates the voice of the person speaking within the listener's field of vision. The result of this is a positive signal-to-noise ratio on a scale that has never been achieved before.

To round it off, the top model in the new Claro range is the first ever to feature the wireless FM technology found in the well-known MicroLink system and thus has all the advantages of wireless communication technology, such as speech transmission over distance in noisy surroundings (conferences, schools, etc.). As a result, the hearing computer has undergone a definitive transition to a communication/hearing computer, which in many situations is indisputably superior to the human ear.

Claims	Prior Art
	Phonak – BTE Elite Article at Page 1 **Perseo – the outstanding features place Perseo in a class of its own** • **Personal*Logic*** makes Perseo the first digital hearing system with personalized automatic operation. **Personal*Logic*** selects the appropriate base program and continually adapts its parameters to suit the needs of the wearer. • **Dual-base automatic** allows processing and features to be separately adjusted for quiet and noisy situations for optimum automatic performance. Adjustments for noise never disrupt adjustments for quiet and vice versa. • **Digital Perception Processing (DPP)** with Clarity Component Enhancement: Increased clarity and sound quality also in loud environments. Processing in 20 critical bands. • **Personal System Managers**: Experience Manager, Dynamic Feedback Manager, Occlusion Manager, and Listening Situation Manager for easy and individual adjustment of Personal*Logic*. • **Multi-programs** • **Fast tracking Adaptive digital AudioZoom (dAZ)** to separate actively speech from background noise. • **Fine-scale Noise Canceler** in 20 channels. • **TacTronic program switch** • **Optional remote control** • **FM compatible** • **Wind and weather protection** • **Fashion Color Line** (choice among 12 attractive colors) Phonak – Lewis Article re ML8 Improving Speech Perception in Noise with Directional Microphones and Frequency Modulation (FM) Technology (2002) at Page 2 Personal FM systems have also been shown to improve speech intelligibility in noise (Hawkins, 1984; Fabry, 1994; Crandell & Smaldino, 2000). Past investigations have demonstrated that the utilization of FM technology can improve speech intelligibility in noise by as much as 20–25 dB (see Crandell and Smaldino, 2000). With a personal FM system, the speaker's voice is picked-up via FM wireless microphone located near the speaker's mouth – where the effects of reverberation, distance, and noise are minimal. The FM system converts the acoustic signal to an electrical waveform at the microphone, and the signal is transmitted via FM signal, from the transmitter to the receiver. Both the transmitter and the receiver are tuned to the same transmitting and receiving frequency. At the receiver end, the electrical signal is amplified, converted back to an acoustical waveform and conveyed to the listener.

Claims	Prior Art
	Phonak – Product Lines with Claro at Page 1 Elite  Complete Claro product line Claro is the first hearing instrument that analyzes and responds to auditory stimuli virtually like the healthy human ear. Thanks to the further optimization of Phonak's exclusive signal processing strategies, Claro provide a very high degree of hearing comfort and optimized speech intelligibility, particularly in difficult hearing situations. More about Elite ITE, BTE or BTE with integrated FM receiver instruments Lewis – Speech perception in noise: directional microphones (2004) at Page 428 study participants. These subjects used a Phonic Ear 805 CD BTE hearing instrument that had the capability to switch between omnidirectional and directional microphone modes. The Phonic Ear 441T microphone transmitter and the Phonic Ear 445R FM receiver served as the FM system. Speech Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The Lewis – Speech perception in noise: directional microphones (2004) at Page 437 Finally, this investigation was conducted with just one particular model of hearing aid and one model FM system. Currently, a number of other companies manufacture FM system technology that is compatible with hearing aid technology, including Phonic Ear and AVR Communications Ltd. Additionally, the Phonak Microlink is compatible with hearing aids manufactured by 16 other companies. Since several studies have reported a wide degree of electroacoustic variability with the use of various FM components, it should not be assumed that all brands and models of these devices would produce the same speech perception results obtained in this study (Freeman et al, 1980; Bess et al, 1984; Thibodeau and Saucedo, 1991).

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

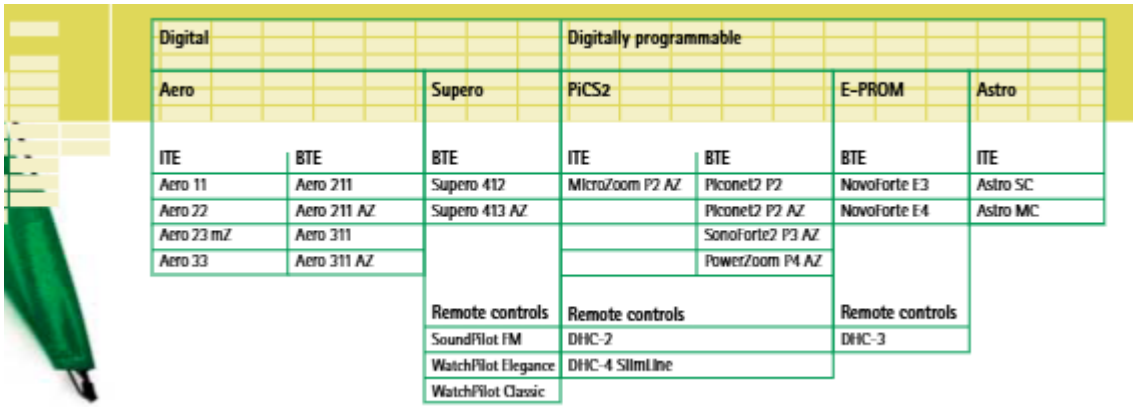
Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 9

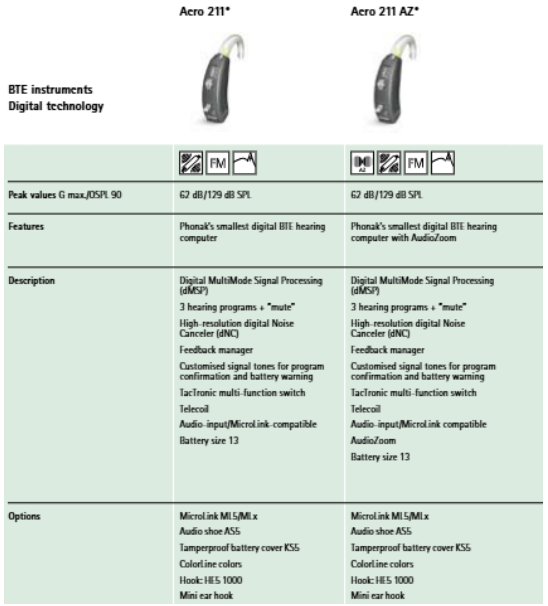
BTE instruments	Claro 111 dAZ Claro 111 dAZ FM	Claro 211 dAZ Claro 211 dAZ FM	Claro 311 dAZ Claro 311 dAZ FM
	FM	FM	FM
Peak values G max./OSPL 90	52 dB/126 dB SPL	62 dB/135 dB SPL	72 dB/135 dB SPL
Features	Specially designed for steep high-frequency and mild hearing losses	Digital speech transparency	Digital speech transparency and higher gain
Description	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (111 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (211 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (311 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection
Options	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)
Programming	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)

Phonak – Collections 2002 Product List at Page 11



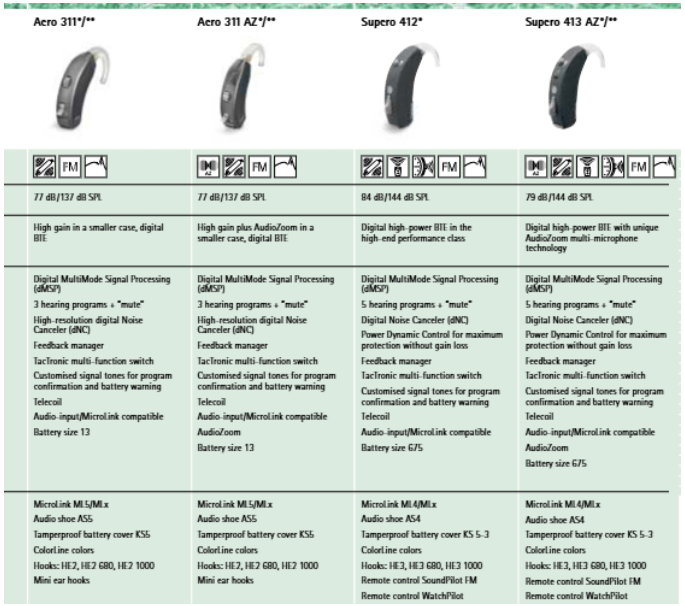
Digital			Digitally programmable			
Aero		Supero	PiCS2		E-PROM	Astro
ITE	BTE	BTE	ITE	BTE	BTE	ITE
Aero 11	Aero 211	Supero 412	MicroZoom P2 AZ	Piconet2 P2	NovoForte E3	Astro SC
Aero 22	Aero 211 AZ	Supero 413 AZ		Piconet2 P2 AZ	NovoForte E4	Astro MC
Aero 23 mZ	Aero 311			SonoForte2 P3 AZ		
Aero 33	Aero 311 AZ			PowerZoom P4 AZ		
		Remote controls	Remote controls		Remote controls	
		SoundPilot FM	DHC-2		DHC-3	
		WatchPilot Elegance	DHC-4 SlimLine			
		WatchPilot Classic				

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 16



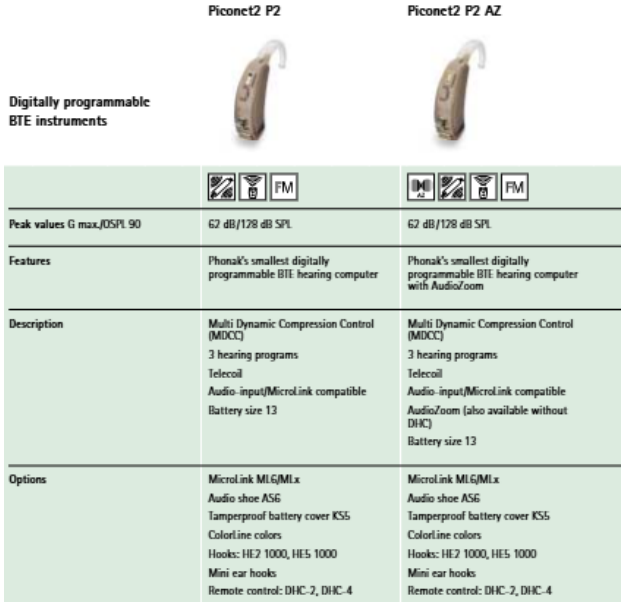
BTE instruments Digital technology	Acro 211*	Acro 211 AZ*
	FM	FM
Peak values G max/OSPL 90	62 dB/129 dB SPL	62 dB/129 dB SPL
Features	Phonak's smallest digital BTE hearing computer	Phonak's smallest digital BTE hearing computer with AudioZoom
Description	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink-compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink compatible AudioZoom Battery size 13
Options	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 17



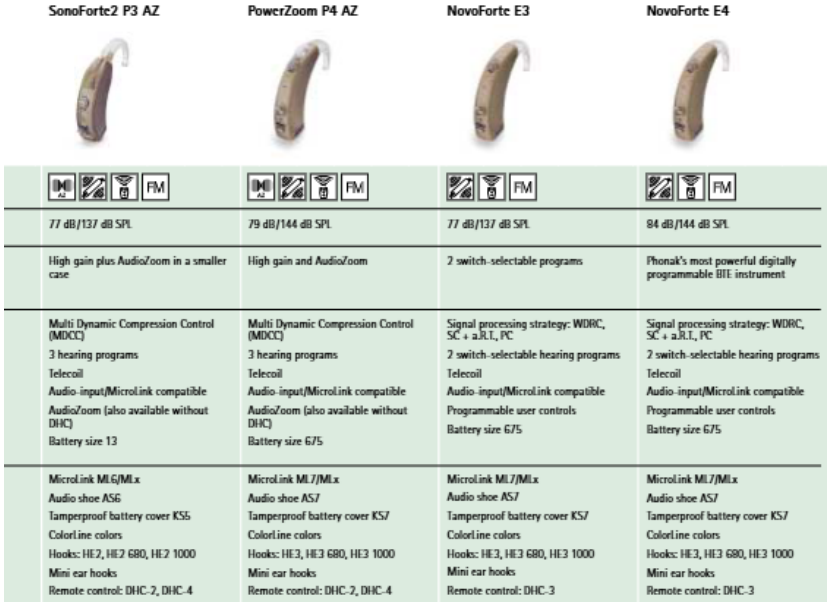
Aero 311*/**	Aero 311 AZ*/**	Supero 412*	Supero 413 AZ*/**
77 dB/137 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL	79 dB/144 dB SPL
High gain in a smaller case, digital BTE	High gain plus Audio/zoom in a smaller case, digital BTE	Digital high-power BTE in the high-end performance class	Digital high-power BTE with unique Audio/zoom multi-microphone technology
Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 13	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 675
MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 18



Digitally programmable BTE instruments	Piconet2 P2	Piconet2 P2 AZ
	FM	FM
Peak values G max/OSPL 90	62 dB/128 dB SPL	62 dB/128 dB SPL
Features	Phonak's smallest digitally programmable BTE hearing computer	Phonak's smallest digitally programmable BTE hearing computer with AudioZoom
Description	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13
Options	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 19



	SonoForte2 P3 AZ	PowerZoom P4 AZ	NovoForte E3	NovoForte E4
	AZ FM	AZ FM	FM	FM
	77 dB/137 dB SPL	79 dB/144 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL
	High gain plus AudioZoom in a smaller case	High gain and AudioZoom	2 switch-selectable programs	Phonak's most powerful digitally programmable BTE instrument
	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675
	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS6 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 21

Value Line features:

- Digital technology as well as proven, modern analog technology, full FM compatibility
- Calibrated settings of the AudioProgrammer switch reproducible at all times
- Superior amplifier power with excellent acoustic quality
- Reliable, easy-to-service instruments



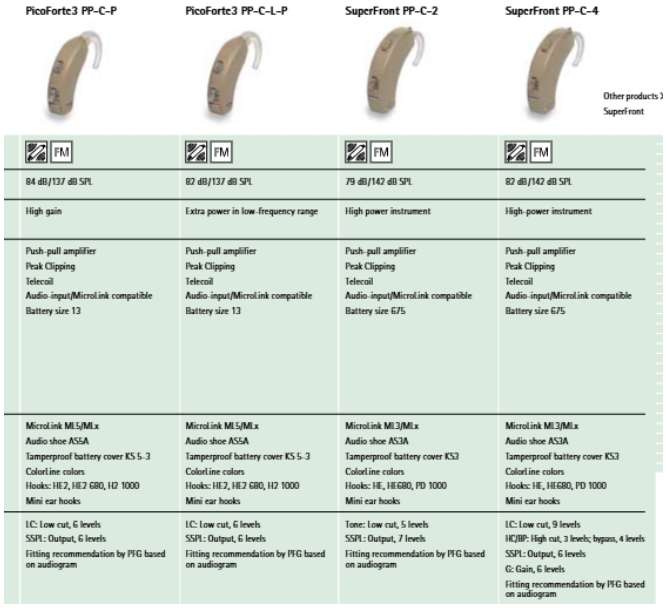
Classica Classica C-D SuperFront

IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 284

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 24

BTE instruments	Supero 411*	PicoForte3 SC-D	PicoForte3 PP-SC
			
	FM	FM	FM
Peak values G max./OSPL 90	79 dB/142 dB SPL	64 dB/127 dB SPL	78 dB/133 dB SPL
Features	Digital high-power BTE	Directional microphone, switchable	High power
Description	dLimiting signal processing strategy 1 hearing program + "mute" Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13	Super Compression + a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13
Options	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, H3 1000	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks
Programming	Programmable with PC (IBM-compatible) and HI-PRO programming interface PFG Software, version 8.0 or above	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 25



PicoForte3 PP-C-P	PicoForte3 PP-C-L-P	SuperFront PP-C-2	SuperFront PP-C-4
			Other products) Superl·ront
FM	FM	FM	FM
84 dB/137 dB SPL	82 dB/137 dB SPL	79 dB/142 dB SPL	82 dB/142 dB SPL
High gain	Extra power in low-frequency range	High power instrument	High-power instrument
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks
LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	Tone: Low cut, 5 levels SSPL: Output, 7 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels HC/BP: High cut, 3 levels; bypass, 4 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 26

	SuperFront PP-C-L-4 SuperFront PP-C-L-4+	SuperFront PP-SC	Classica AGC
BTE instruments			
	FM	FM	FM
Peak values G max./OSPL 90	82 dB/142 dB SPL (PP-C-L-4) 81 dB/142 dB SPL (PP-C-L-4+)	82 dB/140 dB SPL	65 dB/133 dB SPL
Features	High-power low-freq. range (PP-C-L-4) Super low-freq. instrument (PP-C-L-4+)	High-power	AGC-O with a.R.T.
Description	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Broadband response Telecoil Audio-input/MicroLink compatible Battery size 675
Options	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
Programming	LC: Low cut, 9 levels HC: High cut, 6 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels BP/HC: Bypass, 6 levels; high cut, 3 levels SSPL: Output, 9 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 27

Classica C-D	Classica PP-SC	Classica PP-C-P	Classica PP-C-L-P
			
FM	FM	FM	FM
70 dB/139 dB SPL	80 dB/137 dB SPL	84 dB/144 dB SPL	80 dB/144 dB SPL
Broadband BTE with directional microphone	High-power	High gain	Extra power in low-frequency range
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink-compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Phonak – Collections 2002 Product List at Page 29

FM receivers:



Claro dAZ FM
MicroLink receiver directly integrated in the hearing instrument



MLx
Universal MicroLink receiver compatible with most makes of hearing instrument



ML3–ML7
MicroLink receivers compatible with Phonak hearing instruments



MicroVox RX2
Classic robust FM receiver for cochlear implants and induction coils

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 30

MLx, the most flexible FM receiver ever made, can be used not only with Phonak BTEs but also with most third-party hearing instruments available on the market today. This is possible thanks to the standard Euro connector which simply plugs into the audio shoe of the hearing instrument.

Once again, Phonak demonstrates its commitment to innovative developments that serve hearing-impaired adults and children.

MLx compatible BTE makes:

- Argosy
- Audifon
- Beltone
- Danavox
- GN ReSound
- Hansaton
- Interton
- Oticon
- Philips
- Phonak
- Rexton
- Siemens
- Sonic Innovations
- Unitron
- Widex









Detailed, up-to-date compatibility lists of Phonak's MLx MicroLink receiver are available online at:

www.phonak.com/mlx

Phonak – Collections 2002 Product List at Page 31

Phonak FM provides a dramatic improvement of speech intelligibility because the signal is picked up directly at the source (the speaker's voice, the television's audio channel, or the telephone receiver) and is then transmitted without wires to the miniaturised receiver. The receiver (Phonak MicroLink MLx or MicroVox RX2) supplies a high-quality signal directly to the speech processor of the user's cochlear implant.

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 33

PFG –
new perspectives
in fitting success





QuickGuide
The versatile, easy-to-use online tour through the PFG fitting process with access to all optional fitting tools.

Instrument selection wizard
This online assistant recommends the product that best matches the customer's needs.

Initial SoundCheck
This is an ideal tool for fast and simple fine tuning of sound quality. Three controls provide convenient gain and balance settings in the high- and low-frequency ranges.

Automatic fine tuning
Based on the customer's hearing perception, the software automatically computes the most suitable modifications.

Manual fine tuning
A high degree of fitting precision can be attained with this broad spectrum of parameters in an optimised display format.

Phonak Sound
The degree of fitting success can be dramatically improved by using these real-life sound clips. All sound examples can be saved on a computer as MP3 files; they are also available on a sound CD.

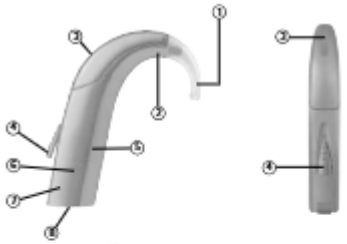
Proven PFG benefits
- One program for all Phonak hearing computers
- NOAH compatible or standalone version
- Detailed help system with in-depth scientific and technical information
- PFG training demo

Phonak – Claro 211 Product Information at Page 1

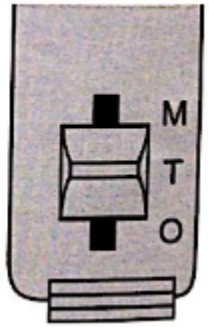
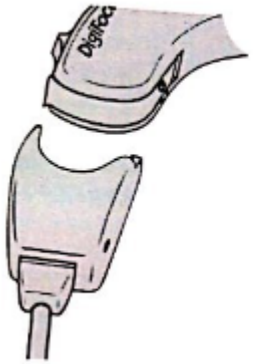
Digital, automatic, BTE hearing instrument with FM module integrated into the streamlined housing. Featuring Phonak's Digital Perception Processing (DPP), three hearing programs and automatic program selection for quiet and for environments with speech in broadband noise. Includes Adaptive digital AudioZoom for optimization of Speech in noise and the Fine-scale Noise Canceler. Manual control via the TacTronic switch or the optional mini remote control SoundPilot FM or the remote control WatchPilot.

Phonak – Claro 211 Product Information at Page 1

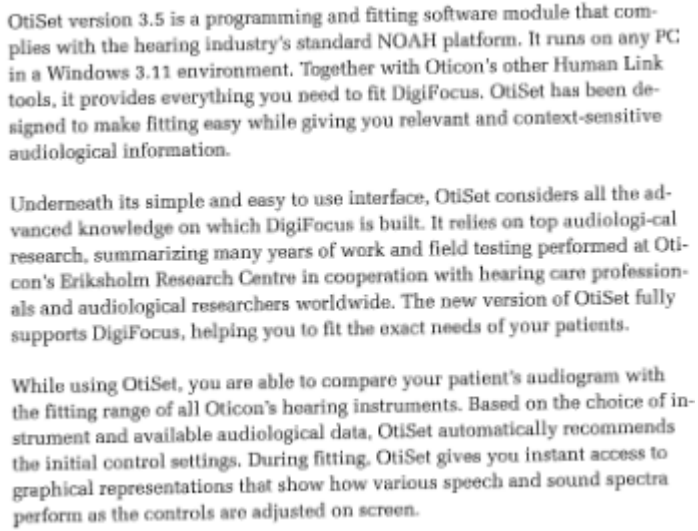
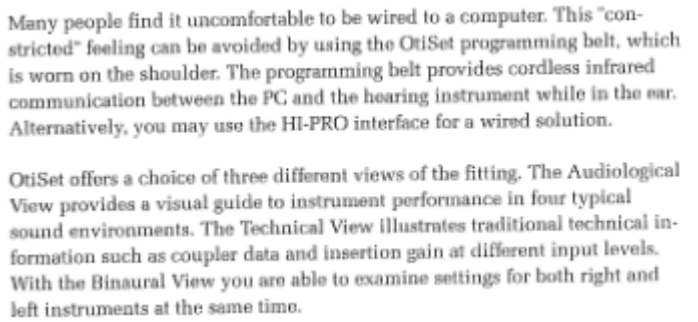
FM features
- World's smallest FM microchip
- Compatible with all Phonak FM transmitters
- Cordless TV and Radio link up
- FM performance corresponding to audiological fitting
- Soft Squelch system
- Built in antenna
- Band H (173–185 MHz); band J (186–201 MHz); band N (207–217 MHz)

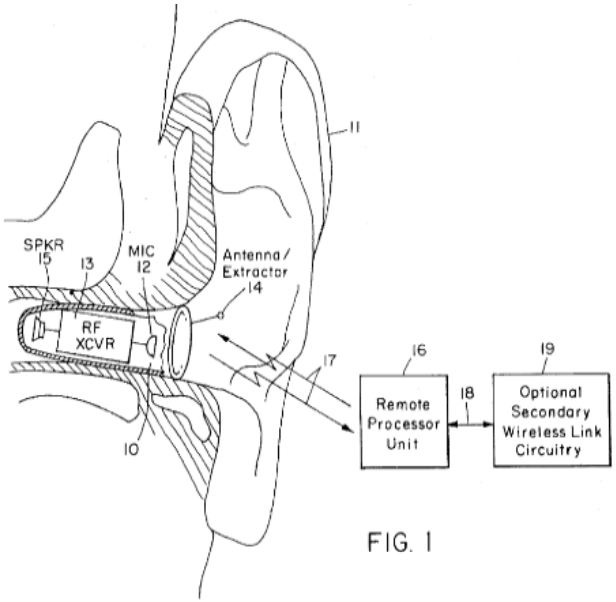
Claims	Prior Art
	Phonak – Claro 211 Product Information at Page 1  ① Ear hook HE4 330 ② Broadband receiver ③ Miniature electret dual micro-phones (protected inputs) ④ TacTronic switch ⑤ FM socket with BTE serial num-ber and integrated program-ming socket ⑥ FM module ⑦ Battery compartment with FM module serial number ⑧ Battery cover combined with optional L/R identification and/or tamperproof system Phonak – Claro 211 Product Information at Page 1 **Processing features** – Digital Perception Processing (DPP) in 20 critical bands – Gain and output controlled by DPP – 3 hearing programs (+ mute mode): • QuietAdapt for hearing in relative quiet • NoiseAdapt for hearing speech in noise • Program 3 telephone default (other choices available) – FM programs: FM and FM+M – AutoSelect: automatic switching between QuietAdapt and NoiseAdapt – Claro's noise suppression technologies • Adaptive digital AudioZoom (dAZ) • Fine-scale Noise Canceler in 20 bands • Soft squelch Phonak – Claro 211 Product Information at Page 2 **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program.

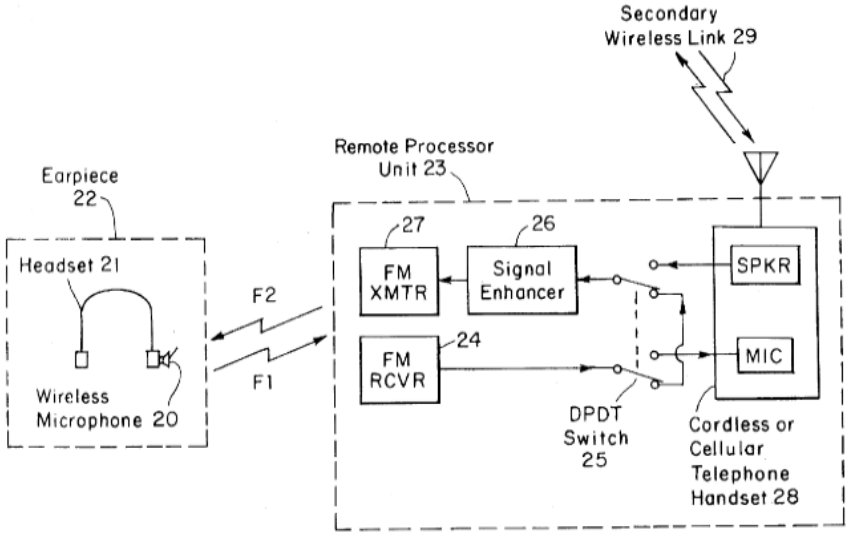
Claims	Prior Art
	Phonak Claro - Technology of the Digital Future at Page 2  Claro - integrated FM receiver FM technology The first hearing computer with an integrated FM receiver. Interference-free communication, even with high levels of noise or echo, or at a distance, are the advantages offered by the design of the Claro series - integrating the smallest and most highly-developed FM system in the world. Phonak Claro – Remote control WatchPilot and Claro dAZ at Page 1 **Claro from Phonak, the first bionic, fully digital hearing computer, sets new standards in hearing instrument technology. At the 45th International Congress of Hearing Aid Acousticians in Cologne (October 5 – 7) the Swiss company, whose name is synonymous with high-tech innovation, presents two more world-firsts: WatchPilot, a wristwatch with an integrated hearing system remote control, and Claro dAZ FM hearing computers, a fully digital hearing system with an integrated miniature radio receiver. At the same time, Phonak will be unveiling Claro 311 dAZ, the most powerful behind-the-ear instrument in the Claro line to date. "The new products represent major advances in hearing instrument technology," says Peter Pfluger, CEO of the Phonak Group. "We are giving hearing impaired individuals an improved quality of life and helping to turn the hearing instrument into a modern communication system."** Phonak Claro – A World First from Phonak at Page 1 **Increased communication capabilities: top models with integrated FM technology** One of the outstanding features of the top-of-the-line Claro BTE (Behind-The-Ear) model is the cordless FM technology found in the well-known Phonak MicroLink system. For the first time ever, this radio receiver has been completely integrated in a hearing system. This means that the instrument now has all the advantages of wireless communication technology, such as speech transmission over distance, in environments with a high level of reverberation and in noise (conferences, schools, etc.). The hearing computer can be turned into a fully-fledged communication system which in many situations is actually superior to the natural, healthy ear.

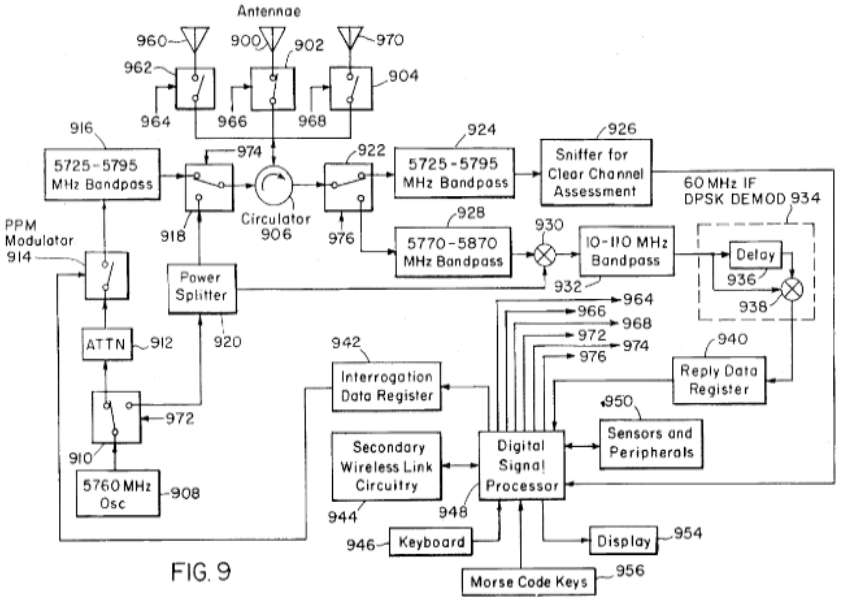
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4 **Function switch** **M:** Hearing instrument switched **on** and microphone connected. **Normal use.** **T:** Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise. **O:** Hearing instrument switched **off.**  Oticon – DigiFocus Compact User Instructional Brochure at Page 6  **Direct Audio Input socket** (optional) Your hearing instruments may have a Direct Audio Input socket (shoe). This facility allows signals from external sound sources to be recieved directly by your hearing instruments. These sources include radio, cassette player, TV, auditory training equipment and assistive listening devices. You will hear the external signal when the function switch is set to the "M" position. Oticon – DigiFocus Compact User Instructional Brochure at Page 11 *5. Telecoil use in church, theatre, or cinema (optional feature)* An increasing number of churches, theatres and public buildings have assistive listening devices installed, usually induction loops. These systems send out wireless sound to be intercepted by the telecoil in your hearing instruments. Ask your hearing care professional for details.

Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 1 DigiFocus is a fully digital hearing instrument built around the digital audio processor, DAP. This processor is designed to have a specific functionality and to comply with a series of physical, acoustical and audiological demands. □ Oticon – DigiFocus White Paper (1996) at Page 4 The input signal from the microphone or the telecoil is digitized and fed to a filterbank, which divides the signal into seven band-pass filters with centre frequencies at: 250, 700, 1200, 2000, 3000, 4000 and 5000 Hz. The filterbank covers the frequency range from 125 to about 6000 Hz. The output from each band-pass filter is equipped with an attenuator by which each filtered signal can be weighted. This allows for a very precise fitting in accordance with the audiometric profile of the hearing impaired. Oticon – DigiFocus White Paper (1996) at Page 9 With DigiFocus you have an extremely flexible hearing instrument. By using the right tools you can define a personal, social and audiological profile for each individual patient. This enables you to utilize the capacity of DigiFocus to the fullest extent. The Human Link tools for DigiFocus include: Oticon – DigiFocus White Paper (1996) at Page 9 • The DigiFocus diary enables you to determine quickly when a hearing instrument needs adjustment • The fine-tuning recommendations help you to adjust the settings • OtiSet, a PC based software program, and the programming belt enable you to fit and fine tune DigiFocus while the client is wearing the instrument. This takes place without your patient being wired to the PC.

Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Page 1-0  OtiSet version 3.5 is a programming and fitting software module that complies with the hearing industry's standard NOAH platform. It runs on any PC in a Windows 3.11 environment. Together with Oticon's other Human Link tools, it provides everything you need to fit DigiFocus. OtiSet has been designed to make fitting easy while giving you relevant and context-sensitive audiological information. Underneath its simple and easy to use interface, OtiSet considers all the advanced knowledge on which DigiFocus is built. It relies on top audiologi-cal research, summarizing many years of work and field testing performed at Oticon's Eriksholm Research Centre in cooperation with hearing care professionals and audiological researchers worldwide. The new version of OtiSet fully supports DigiFocus, helping you to fit the exact needs of your patients. While using OtiSet, you are able to compare your patient's audiogram with the fitting range of all Oticon's hearing instruments. Based on the choice of instrument and available audiological data, OtiSet automatically recommends the initial control settings. During fitting, OtiSet gives you instant access to graphical representations that show how various speech and sound spectra perform as the controls are adjusted on screen. Oticon – DigiFocus White Paper (1996) at Page 11  Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time. Anderson ’783 at Abstract A hearing aid or audio communication system includes an earpiece (10) that can be hidden in the ear canal, and which communicates wirelessly with a remote processor unit, or RPU (16), that enhances audio signals and can be concealed under clothing. Sounds from the environment are picked up by a microphone (12) in the earpiece and sent with other information over a two-way wireless link (17) to the RPU (16). The wireless link (17) uses microwaves for component miniaturization. Furthermore, use of radar technology to implement the wireless link (17), with an RPU (16) interrogator and earpiece (10) transponder, reduces earpiece size and power, as no microwave oscillator is needed in the earpiece (10).

Claims	Prior Art
	Optional secondary wireless link circuitry (19) can be used between the RPU (16) and a cellular telephone system or other sources of information. Electronic voice recognition and response can control system operation. Anderson ’783 at FIG. 1  FIG. 1

Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2

Claims	Prior Art
	Anderson ’783 at FIG. 9  FIG. 9 Anderson ’783 at 1:5–18 The present invention is directed to hearing aids, and in particular to a hearing aid having an earpiece housing worn in or at the ear and a remote processor unit (RPU) worn by or located near the user that wirelessly receives signals from and transmits signals to the earpiece. The present invention is also directed to the use of hearing aid systems as wireless communication devices for hands-free cellular telephone and mobile radio communication "handsets," covert operation and control of hearing aids, hearing protection and noise cancellation simultaneous with binaural hearing aid functions, hearing test equipment, location of misplaced hearing aid system components and wireless cochlear implants. Anderson ’783 at 1:50–65 The present invention uses an audio path that is different from the audio path used by devices in the prior art. The present invention uses a primary two-way wireless link (not a one-way link as in the prior art) between an earpiece worn in or at the user's ear and an RPU worn by or located near the user. In the present invention, audio signals from the environment are picked up by a microphone in the earpiece (not a hand-held microphone as used by wireless hearing aids in the prior art) and transmitted over the primary two-way wireless link to the RPU (instead of going to processor electronics contained in the earpiece, as

Claims	Prior Art
	in basic hearing aids known in the prior art), where the audio signals are enhanced according to the user's needs before transmission over the primary wireless link to the earpiece. Signal processing is performed in the RPU rather than the earpiece to take advantage of relaxed size and power constraints. Anderson ’783 at 2:20–39 It is an object of the present invention to provide a new and useful auditory aid for hearing-impaired persons (i.e., those having certain residual hearing) by removing audio signal enhancement functions from the earpiece and placing them in an RPU. Use of an RPU provides several advantages over systems that attempt to place all system capabilities within the earpiece. The RPU approach allows a simple earpiece design comprising a miniature low-power wireless transceiver, microphone and speaker. Note that the speaker is also known in the hearing aid art as a "receiver," but the term "speaker" is used here to avoid confusion (similarly, the term "talker," not "speaker," is used to describe a human producing vocal sounds). The resulting earpiece is extremely small, can be hidden from view in the ear canal if desired, and allows complete freedom of movement when a primary two-way wireless link to the RPU is used. Processing for all major system capabilities, such as amplification and other forms of signal enhancement, takes place in the RPU where size and power constraints are relaxed, leading to a cost-effective design. Anderson ’783 at 3:13–18 FIG. 1 is a block diagram system overview illustrating how an earpiece (comprising a microphone, RF transceiver, dual-use antenna/extractor and speaker) is worn in the ear, communicates via a primary two-way RF link with an RPU, and also communicates via an optional secondary wireless link to a telephone system. Anderson ’783 at 3:19–21 FIG. 2 is a block diagram illustrating a limited-feature preferred embodiment of the invention that uses commercial off-the-shelf system elements. Anderson ’783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the

Claims	Prior Art
	dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 4:30–33 Although the system described here is monaural, it will be clear to those skilled in the art that a binaural (stereo) system can be constructed based on the same principles as the monaural system. Anderson '783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson '783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and

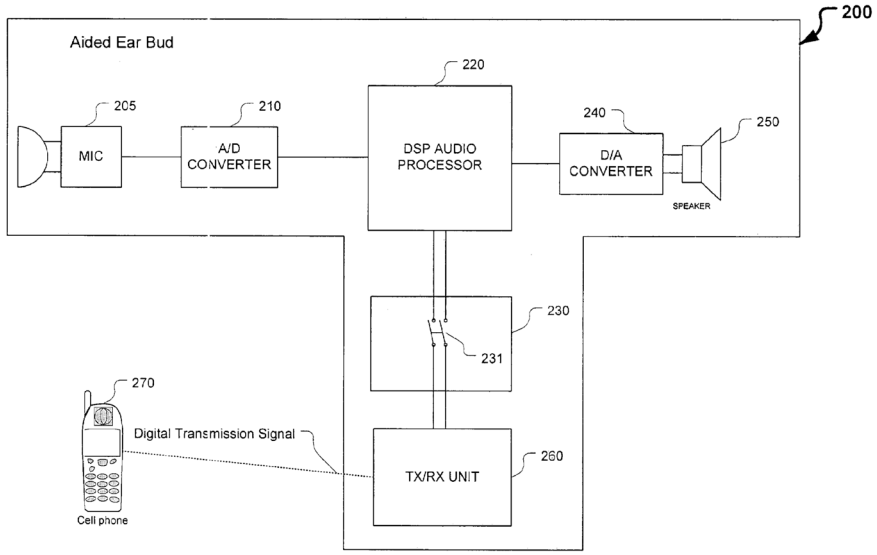
Claims	Prior Art
	transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 12:20–32 During the reply interval 52, a modulated (via modulator 830) low-power 60 MHz oscillator 835 is applied to voltage point V1, causing the diode 43 to turn on and off at a 60 MHz rate and act as a mixer. The mixer diode 43 generates frequency components centered at the difference frequency of 5700 MHz and the sum frequency of 5820 MHz. The 5820 MHz sum component passes through the resonator 42, and forms the reply which is wirelessly transmitted from the earpiece transponder antenna 40 to the RPU reply processor circuitry. The earpiece 60 MHz oscillator 835, which is only turned on during the reply interval, changes phase at specified points in time to generate the DPSK reply data. Anderson ’783 at 12:47–13:22 Remote processor unit interrogator and reply processor: The RPU, shown in FIG. 9, may use any of a wide variety of components well known in the art. The DSP 948 may contain several integrated circuits, e.g., the Motorola DSP56L002 single-chip digital signal processor, a Motorola DSP56200 cascadable-adaptive digital filter chip and supporting memory devices well known in the art. Miniature components such as dielectric resonance oscillators are available for the microwave oscillator 908, and the DPSK replies can be decoded using surface acoustic wave (SAW) or glass bulk acoustic wave (BAW) delay lines 936 in a standard delay-and-multiply demodulator circuit 934. An alternative demodulator that achieves superior noise immunity can be implemented using a phase-locked loop in a squaring loop configuration for carrier recovery instead of delay-and-multiply circuitry 934. Decoding of a DPSK waveform using such coherent detection techniques is known in the art as differentially encoded coherent phase shift keying, or DECPSK. Amplifiers using low noise pseudomorphic high electron mobility transistors (e.g., the HP ATF-36077) may be used as needed (e.g., between the circulator 906 and routing switch 922), and such amplifiers are not individually identified in FIG. 9. All switches shown in FIG. 9 are RF switches that can be implemented using, e.g., PIN diodes. Switch control connections from the DSP 948 to the RF switches are shown in FIG. 9 for clarity, and all switches remain set as shown until changed as noted. The system shown in FIG. 9 demodulates a received reply when the RF reply travels

Claims	Prior Art
	through a selected antenna 900, antenna selection switch 902, circulator 906, routing switch 922 (set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 976), bandpass filter 928 and IF mixer 930. The IF mixer 930 downconverts the reply by mixing it with a small amount of RF power from the 5760 MHz oscillator 908 via switch 910 (which is set to the position opposite that shown in FIG. 9 during the reply interval by DSP 948 via control signal 972) and power splitter 920. The resulting DPSK reply has a center frequency of 60 MHz, a bandwidth of 100 MHz, and is passed through the bandpass filter 932 to the DPSK demodulator 934 which, in this example, consists of a 50 nsec delay line 936 and multiplier 938. The resulting bits are stored in the reply data register 940 for processing by the DSP 948. Anderson ’783 at 21:40–46 If no interrogation is received by the earpiece 10 from the RPU 16 for a certain period of time (e.g., 0.1 second), as determined for example by a retriggerable monostable multivibrator circuit (also known in the art as a "one-shot") implemented in the earpiece ASIC 845, the ASIC initiates simulation of a stand-alone CIC hearing aid earpiece. Anderson ’783 at 21:46–50 Stand-alone operation is provided when the ASIC 845 causes signals from the microphone 875 via the 8-bit compressing A/D 880 to be routed to the speaker 855 via the 8-bit expanding D/A 850. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of

Claims	Prior Art
	wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other

Claims	Prior Art
	information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 24:47–66 Morse code keys 956 or voice recognition capabilities implemented using the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. In this respect, the system acts as a "virtual handset" communication aid that replaces a conventional telephone handset in many applications (although users may additionally communicate with a conventional handset or speakerphone in the usual fashion). To accomplish the virtual handset function, the RPU 16 is equipped with optional secondary two-way wireless link circuitry 19 to communicate with a telephone or other communication network. The secondary wireless link is separate from, and does not interfere with, the primary wireless link 17 between the earpiece 10 and RPU 16. In applications where relatively high-power transmissions take place on the secondary link and the secondary link transmitter is integrated into the RPU, the RPU can be placed on a user's wrist or ankle away from vital organs to reduce any potential health risk relative to conventional wireless handsets. Anderson ’783 at 27:4–24 Hearing test: The present invention allows hearing tests to be performed via the earpiece normally worn by the user, without the need for additional expensive test equipment. All signal generation capabilities necessary to perform hearing tests are available in the hearing aid system comprising an earpiece 10 and RPU 16. Specifically, the RPU DSP 948 can be used to generate tones of varying frequencies and

Claims	Prior Art
	amplitudes, as well as other signals known to be useful for hearing test purposes (e.g., nonsense syllables) that are converted to acoustic waves by the earpiece. Such tones are used to perform a hearing test that determines appropriate gain vs. frequency parameters for a program stored in the RPU DSP 948 that performs signal enhancement to compensate for the user's hearing loss. Note that the hearing test program that controls the RPU DSP 948 during the hearing test can be temporarily stored in the RPU DSP 948 for the duration of the test, then deleted upon completion of the test to allow re-use of RPU DSP 948 memory resources during normal operation. The hearing test program may be loaded into the RPU DSP 948 through the secondary wireless link 944 or a wired peripheral link 950. Enzmann ’730 at Abstract An aided ear bud device is provided that operates as a hearing aid device as well as an ear bud device for a wireless phone. The aided ear bud normally operates as a hearing aid device. The aided ear bud state is changed by a verified input from a wireless phone. Upon the change in state, the microphone input from the aided ear bud is remotely output to the wireless phone while the incoming transmission from the wireless phone is combined with the microphone input and output to the user. The ear bud device can be modified to receive analog or digital inputs from other outside sources.

Claims	Prior Art
	Enzmann ’730 at FIG. 2  FIG. 2 Enzmann ’730 at 1:7–11 This invention relates to systems and methods for providing a hearing aid capable of communicating with a wireless phone. More particularly, this invention provides a hearing aid capable of simultaneously or alternatively performing the functions of an ear bud for the wireless phone. Enzmann ’730 at 2:17–23 Therefore, this invention provides systems and methods for adapting a hearing aid to operate as an ear bud device in addition to providing conventional hearing aid functions. Using this method, this invention is able to address many of the issues hearing impaired wireless phone users encounter while addressing the inherent problems associated with the conventional solutions. Enzmann ’730 at 3:20–34 The conventional hearing aid comprises a microphone 105, an A/D converter 110, a digital signal audio processor (DSP) 120, a D/A converter 130 and a speaker 140. In operation, sounds, represented by an analog signal and/or ambient noise emitted from an outside source, is captured by the microphone 105 and converted to a digital signal by the A/D converter 110. The DSP 120 manipulates the converted signal in

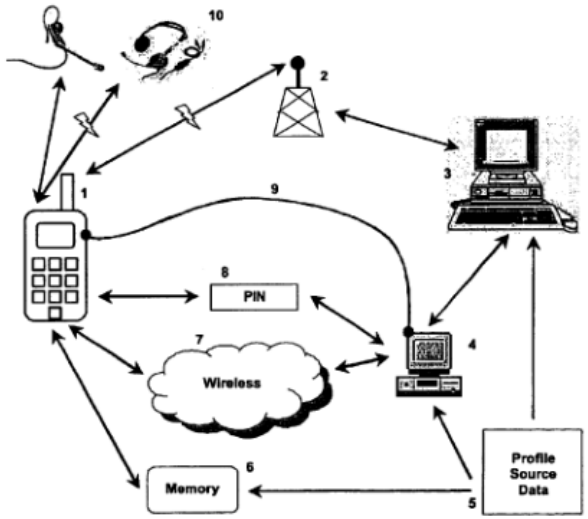
Claims	Prior Art
	accordance with well-known techniques and sends a processed signal to the D/A converter 130. In the D/A converter 130, the processed signal is converted to an analog signal and output to the speaker 140. The speaker 140 is placed in or around the hearing impaired user's ear. Thus, the captured sounds have been captured, processed, and amplified and output to the user. Enzmann '730 at 3:35–40 The manipulating processes of the DSP 120 serves both to amplify the converted signal from the A/D converter 110 and correct any degraded signal areas so that an improved signal is sent to the speaker 140. The manipulation processes may also adjust the incoming signal based on a preprogrammed frequency mapping of the user's hearing characteristics. Enzmann '730 at 3:46–49 The embodiments of the present invention allow the user to hear the audio transmissions of the wireless phone in the speaker of the aided ear bud. This is achieved by transmitting the audio output to the aided ear bud. Enzmann '730 at 3:55–57 In addition, the portability of the wireless phone is increased because the user's hearing device can now act as a wireless ear bud for the wireless phone. Enzmann '730 at 4:14–16 In the second operating mode, the TX/RX unit 260 transmits and receives communications from the wireless phone 270. Enzmann '730 at 4:26–29 Ambient noise and the user's voice are captures by the microphone 205 and processed and combined with signals from the TX/RX unit 260 in the DSP 220 for output through the D/A converter 246 and the speaker 250. Enzmann '730 at 4:43–50 The connection shown between the TX/RX unit 260 and the wireless phone 270 is preferably a wireless digital connection, such as Bluetooth, etc. However, it should be appreciated that the connection between the TX/RX unit 260 and the wireless phone 270 can be any kind of wireless or wired connection, and may

Claims	Prior Art
	be digital or analog, infrared (IR) or transmitted radio frequencies (RF) without, departing from the scope of the invention. Enzmann ’730 at 4:63–5:5 Signals received at the microphone 205 are converted by the A/D converter 210, manipulated by the signal-processing function 221 which outputs 226a is then passed to the op amp 222. The output 226 and 226b of the op amp 222 are electronically connected to the TX/RX unit 260 and to the op amp 223. The output 226 connected to the TX/RX unit 260 is then output to the wireless phone 270. The signal from the TX/RX unit 260 is used by the wireless phone 270 as if the user was speaking into the microphone of the wireless phone. Enzmann ’730 at 5:13–19 The signals received at the microphone 205 are passed to the user as background noise and must be passed to the wireless phone 270 as if the user was speaking directly into the wireless phone 270. Remotely transmitting this signal to the wireless phone from the aided-ear bud 200 also allows the user to be located in a remote location away from the wireless phone 270. Enzmann ’730 at 5:20–30 The signal 226b passed to the op amp 223 is representative of ambient noise in the aided ear bud 200. This signal 226b is typically reduced in volume and combined with the input from the signal-processing function 225 to produce a mixed signal. The signal processing function 225 processes signal 228 from the TX/RX unit 260 and provides a second processing step so that the timing of the incoming signals from the microphone and the wireless phone are in sync and can be effectively combined to create a true representation of sound to the user. The op amp 223 outputs the mixed signal to the D/A converter 240, and finally to the speaker 250. Enzmann ’730 at 6:12–35 The amplification chart 325 contains a series of pre-stored dynamic frequency maps based on the user's hearing characteristics, the speaker characteristics and the ambient noise near the aided ear bud 200. The digital logic 320 will receive the transmitted signal from the TX/RX unit 260 and process those signals to an optimal form based on the user's hearing, speaker and ambient noise characteristics. The arrangement of the logic devices within the digital logic 320 will serve to produce each of the characteristics (wave shape) of the amplitude chart 325. The user's hearing characteristics remain static unless the aided ear bud

Claims	Prior Art
	200 is reprogrammed to compensate for additional hearing loss or increase and different users. For example, as shown in FIG. 5, a mapping of a user's hearing characteristic (denoted by the "••" line) illustrates that the user lacks significant hearing capability in higher frequencies. The desired hearing characteristic is denoted by the "-••" line. This desired hearing characteristic depicts or represents how a non-hearing impaired person's hearing characteristics may be. As known to those skilled in the art, close representation of this hearing characteristic is achievable by adding a complementary characteristic to the user's hearing characteristic to produce a result that is very close to the desired characteristic. Each of the characteristics is corrected using the process shown in FIG. 5. Enzmann '730 at 7:34–45 In this configuration, the A/D converter 410 converts incoming analog signals into a digital signal for use by the aided ear bud in accordance with the above-description and converts digital output signals to analog format and output to devices. This configuration allows a user to attach any analog device (or analog output from a digital device) to the aided ear bud, such as portable electronics (CD players, PDA devices) and any other devices that output analog signals. Otherwise, the operation of the second embodiment is similar to the previously described embodiment once the incoming signal is converted from an analog signal to a digital signal. Enzmann '730 at 8:16–23 At step S600, the aided ear bud retrieves the transmitted signal from the TX/RX unit and then proceeds to step S700. In step S700, the process fixes the transmitted signal according to signal stored in the amplification chart. Once the signal has been fixed and verified, the process proceeds to step S800, where this fixed signal is forwarded across the digital switch and fixed by the DSP. The process then continues to step S900. Jannard '526 at 8:45–47 40. Alternatively, the audio device 10A can include a wireless transceiver for receiving digital signals from another device. Jannard '526 at 18:1–22 The microphone 75 can be disposed anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″. Optionally, the microphone 75 can be in the form of a bone conduction microphone. As such, the microphone 75 is disposed such that the when a user wears the

Claims	Prior Art
	audio device 10A′, the microphone 75 is in contact with the user's head 18. For example, but without limitation, the microphone can be positioned anywhere on the anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″ such that the microphone contacts the user's head. More preferably, the microphone 75 is positioned such that it contacts a portion of the user's head 18 near a bone, such that vibrations generated from the user's voice and traveling through the bone, are conducted to the microphone. In another alternative, the microphone 75 can be configured to be inserted into the meatus 24 (FIG. 2) of the ear canal of the user. Thus, in this modification, the microphone 75 can be substituted for one of the speakers 14, 16. Alternatively, an ear-canal type bone conduction microphone can be combined with a speaker so as to provide two-way communication with the user through a single ear canal. Jannard ’526 at 20:57–64 The interface device 110 includes a power source 112, a transceiver 114, an interface 116, and an antenna 118. The power source 112 can be in the form of disposable or rechargeable batteries. Optionally, the power source 112 can be in the form of solar panels and a power regulator. The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. Jannard ’526 at 21:35–39 The interface 116 can be configured to receive signals from the transceiver 114 that are in the form of digital or analog audio signals. The interface 116 can then send the audio signals to the speakers 14C, 16C through speaker lines 120, 122, respectively, discussed in greater detail below. Jannard ’526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204.

Claims	Prior Art
	As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user.

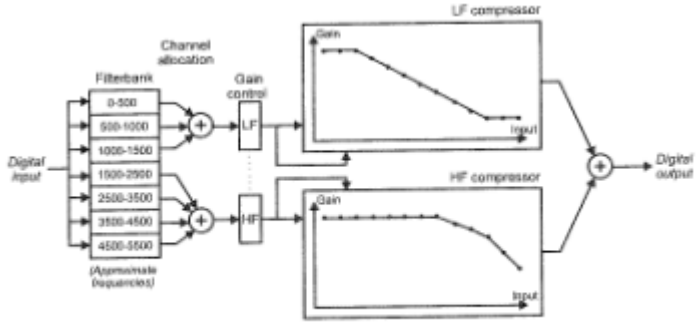
Claims	Prior Art
	Rader ’474 at FIG. 3  FIG. 3 Rader ’474 at 1:26–31 A variety of uses for hearing profiles, other than for the purposes of prescribing hearing aids and assistive listening devices, is being developed. For example, hearing profiles of individuals can be utilized for producing customized audio products, such as pre-recorded music that has been modified according to the hearing profile of the listener. Rader ’474 at 2:1–13 Mobile phones are inherently personal communication devices. Innovations in microelectronics have allowed mobile phones to become extremely portable and wearable, affording access to telecommunications services wherever the service infrastructure is available. Future generation mobile phones may also incorporate entertainment features such as streaming audio and video, playback of stored audio or video, radio reception and presentation, and many other features that deliver audio to the user. It is desirable to enhance the reception of mobile phone system audio by the user, accommodating personal hearing needs, personal choice, or optimization to the noise environment surrounding the user.

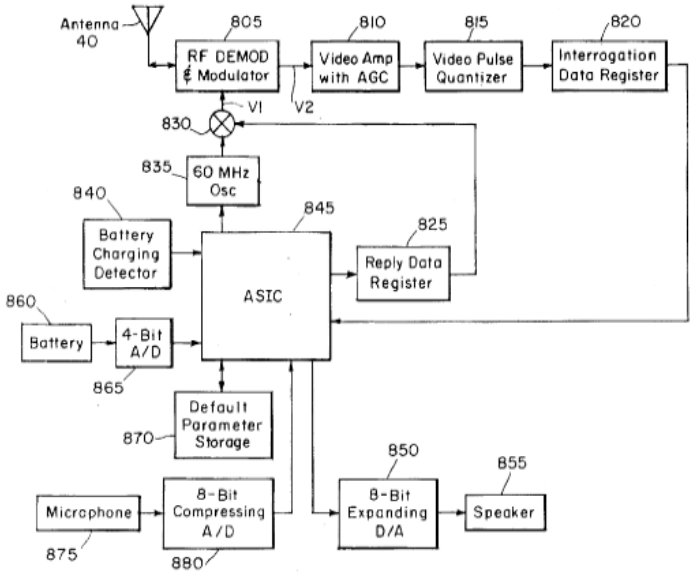
Claims	Prior Art
	Rader ’474 at 2:25–30 According to the present invention, a mobile phone or other personal communication device includes resources applying measures of an individual's hearing profile, personal choice profile, and induced hearing loss profile, separately or in combination, to build the basis of sound enhancement. Rader ’474 at 2:31–45 A personal communication device according to the present invention comprises a transmitter/receiver coupled to a communication medium for transmitting and receiving audio signals; control circuitry that controls transmission, reception and processing of call and audio signals; a transducer such as an internal speaker, an external speaker, and a speaker in an attached headset; and an internal microphone, an external microphone, and a microphone in an attached headset. The control circuitry includes logic applying one or more of a hearing profile of the user, a user preference related hearing, and environmental noise factors in processing the audio signals. In various embodiments of the invention, the control circuitry includes instruction memory and an instruction execution processor such as a digital signal processor. Rader ’474 at 8:7–16 Accessories 10 to the mobile phone may be used in conjunction with the mobile phone and any of the aforementioned subsystems. Accessories 10 may include, for example, monaural or binaural headphones worn by the user. The binaural headphones would allow for gathering of a hearing profile for each ear, which may become more prevalent as mobile phones become stereo audio playback devices. The accessories 10 could communicate with the mobile phone either through direct wire link or wireless link, as shown in FIG. 3. Rader ’474 at 8:52–58 Accessories 20 to the mobile phone 11 may be used to sense noise, calculate an induced hearing impairment, or even provide computational resources to customize audio. Accessories 20 such as headsets may be used to sense the local noise environment at the user's ear, and send that information to the mobile phone 11 either via a wire link or a wireless link, as shown on the diagram. Rader ’474 at 9:49–51 Customized audio data could be presented to the user through the mobile phone or through accessories 30

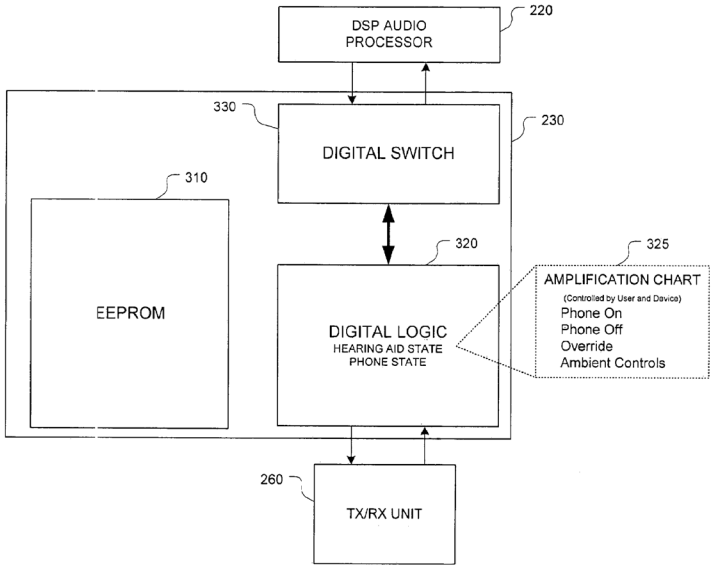
Claims	Prior Art
	such as headsets. Rader '474 at 10:6–12 Customization of audio data can be accomplished by using systems remote from the mobile phone, and delivering the customized or personalized audio to the mobile phone for playback. The alternative is to deliver unprocessed audio to the mobile phone, and use the resources of the mobile phone or an accessory to personalize the audio. A POSITA would have been motivated to combine the DHA system with the features of Anderson, Enzmann, Jannard, or Rader to meet the claim limitations with reasonable expectation of success. For example, a POSITA would have understood the disclosures of the Phonak system to teach well–known structures and functionality associated with a digital hearing aid device, including storing a hearing profile/characteristics for the user based on the results of the hearing test, i.e., such that the hearing profile/characteristics stores the appropriate hearing parameters for the user on the ear–level device itself, and using the hearing profile/characteristics to compensate for the user's hearing loss, as described above in the citations. A POSITA would have found it obvious to configure and use an external device, such a mobile communication device, with the functionality necessary to store a user's hearing profile, execute algorithms for customizing audio according to that stored hearing profile, and transmit the processed audio back to the hearing aid device, in order to leverage the external device's existing communication hardware and processing capabilities to expand the digital hearing aid's functionality to audio signals generated or received by the external device itself, before or instead of signals received by the digital hearing aid itself. To the extent the specific details of features/functionality of similar digital hearing aids are not expressly disclosed, a POSITA thus would have been motivated to look to other references to supply these missing details. A POSITA would have understood that enabling the mobile communication device with such processing capabilities would the reduce power burden and antenna requirements of the hearing aid device and provide an improved user experience by reducing signal drop outs that users frequently experience with signal being transmitted from the external device for processing by the digital hearing air. A POSITA would have understood that this approach would eliminate the need for large and power-consuming electronics in the earpiece. A POSITA would have had a reasonable expectation of success in implementing this combination, as wireless communication between host and ear–level devices and digital signal processing (on the ear–level device for customization according to a user's hearing profile and on an external host device for enhancement and acoustic feedback control) were well–established practices at the relevant time. The combination therefore would have required no more than the routine application of known engineering techniques and would have predictably achieved the intended result to configuring the remote device with functionality already existing on the digital hearing aid.

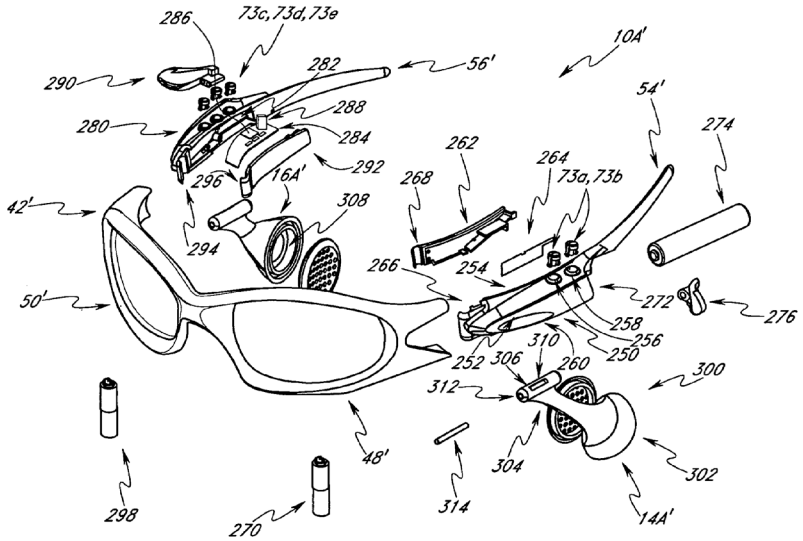
Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Radar, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[17c] electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker,	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker." Phonak – Perseo Brochure at Page 22 

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The Oticon – DigiFocus Compact User Instructional Brochure at p. 2 The programming is accomplished via computer software (OtiSet) using the NOAH standard platform. Additionally, DigiSound can be programmed from Oticon's dedicated programming box, "Easy fit". Oticon – DigiFocus White Paper (1996) at Page 1 The OtiSet software is the single most important tool for fitting DigiFocus. Underneath its simple and easy to use interface, OtiSet sums up all of the advanced knowledge on which DigiFocus is built. By analyzing all of the available personal data, OtiSet automatically recommends the most appropiate settings. In short, OtiSet enables you to draw on all the audiological knowledge available from Oticon. □

Claims	Prior Art
	Oticon – DigiFocus White Paper (1996) at Fig. 1  *Fig.1. DigiFocus - block diagram of the digital audio processor, DAP.* Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 DigiFocus operates on two tiny new computer chips that have the processing power of a normal desktop computer. Traditional hearing instruments have only up to 10 adjustable parameters, but Oticon representatives say that their processor will give a maximum of 100 software-controlled adjustment parameters. And the device can be programmed by an audiologist while the hearing-impaired person is wearing it. Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market.

Claims	Prior Art
	Anderson ’783 at FIG. 8  FIG. 8 Anderson ’783 at 3:40–41 FIG. 8 is a block diagram showing details of an earpiece for a full-featured preferred embodiment of the invention. Anderson ’783 at 12:38–46 Note that all earpiece functions (e.g., power control for the 60 MHz oscillator 835) are controlled by a custom application-specific integrated circuit (ASIC) 845. The ASIC 845 may be a low-power low-voltage device similar, e.g., to the ECI Semiconductor P576 custom programmable linear/digital gate array. Many of the functions that are shown as separate blocks for clarity in FIG. 8, e.g., reply data register 825, may be implemented within the ASIC 845. Anderson ’783 at 21:46–50 Stand-alone operation is provided when the ASIC 845 causes signals from the microphone 875 via the 8-bit compressing A/D 880 to be routed to the speaker 855 via the 8-bit expanding D/A 850.

Claims	Prior Art
	Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Enzmann ’730 at FIG. 4  FIG. 4 Enzmann ’730 at 5:41–44 FIG. 4 provides an expanded view of the source control module 230 according to this invention. The source control module 230 preferably includes at least a digital switch 330, digital logic 320 and a memory device 310. Enzmann ’730 at 5:59–67 The digital switch 330 operates as a double pole, two-position switch. The digital switch 330 is controlled

Claims	Prior Art
	by an input from the digital logic 320. Upon receipt of a proper input from the digital logic 320, the activation state, and thus the position, of the digital switch 330 is changed. The activation state of the digital switch 330 is either opened (OFF) or closed (ON), the latter of which permits input signals to be sent to/from the digital logic 320 to the DSP Audio processor 220. Enzmann '730 at 6:1–11 The digital logic 320 contains a series of devices consisting of multiple identical and separately identifiable structures built of some mix of combinational logic and sequential elements the sequential elements being commonly referred to as registers, latches, flip-flops, or shift register latches. These devices individually or in combination implement a series of activation states and wave patterns that determine the activation state of the aided ear bud and shapes of the frequency wave output from the digital logic 320. The functionality of the digital logic 320 is depicted in FIG. 4 as the amplification chart 325. Jannard '526 at FIG. 3J  *FIG. 3J* Jannard '526 at 3:56–58 FIG. 3J is a front, top, and left side perspective and exploded view of the wearable audio device illustrated

Claims	Prior Art
	in FIG. 3D. Jannard ’526 at 8:38–42 The audio device 10A can be used as an audio output device for any type of device which provides an audio output signal. The audio device 10A can include an audio input terminal disposed anywhere on the eyeglass 40 for receiving a digital or analog audio signal. Jannard ’526 at 10:51–57 Further, some wearers of the audio device 10A′ may prefer to press the speakers 14A′, 16A′ into contact with the outer surfaces of their ears. For example, some users may desire to experience to loudest possible volume from the speakers 14A′, 16A′. Thus, by pressing the speakers 14A′, 16A′ against their ears, the perceived volume of the sound emitted from the speakers 14A′, 16A′ will be the greatest. Jannard ’526 at 12:62–13:6 A cover member 262 cooperates with the housing 250 to define the closed internal cavity 252. In the illustrated embodiment, the internal cavity 252 includes at least one compartment configured to receive an electronic circuit board 264 which includes at least one switch for each of the buttons 73a, 73b. In an exemplary but non-limiting embodiment, the board 264 can include two switches, one for each of the buttons 73a, 73b, which are configured to control a volume output from the speakers 14A′, 16A′. The cover 262 can be attached to the ear stems 54A′ with any type of fastener, such as, for example, but without limitation, screws, rivets, bolts, adhesive, and the like. Jannard ’526 at 13:50–55 The internal cavity 282, in the illustrated embodiment, is configured to receive a printed circuit board 284. In the illustrated embodiment, the printed circuit board 284 includes one switch for each of the buttons 73c, 73d, and 73e. Additionally, the printed circuit board 284 includes an audio file storage and playback device 286. Jannard ’526 at 14:6–14 Advantageously, the printed circuit board 284 also includes or is in electrical communication with a data transfer port 388. In the illustrated embodiment, the housing 280 includes an aperture (not shown) disposed in a position similar to the position of the aperture 272 on the housing 250. In the housing 280, however, the aperture is aligned with the data transfer port 288. Thus, when the printed circuit board 284

Claims	Prior Art
	is received in the internal cavity 282, the data transfer port 288 is aligned with the aperture. Jannard ’526 at 15:60–16:2 Thus, with reference to FIGS. 3J and 3M, the ribbon connector 318 can extend from the housing 280, into the recesses 294, 296, through an aperture (not shown) in the hinge pin 298 to the upper opening within the hinge pin 298, then through the channel 316 (FIG. 3E), to an upper opening of the hinge pin 270, out through an aperture (not shown) through a side of a hinge pin 270, through the recesses 266, 268 of the housing 250, and then to the speaker 14A′, printed circuit board 264, and the power storage device 274. Jannard ’526 at 18:1–22 The microphone 75 can be disposed anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″. Optionally, the microphone 75 can be in the form of a bone conduction microphone. As such, the microphone 75 is disposed such that the when a user wears the audio device 10A′, the microphone 75 is in contact with the user's head 18. For example, but without limitation, the microphone can be positioned anywhere on the anywhere on the frame 40A′, including the orbitals 48A″, 50A″, the bridge 52A″, or the ear stems 54A″, 56A″ such that the microphone contacts the user's head. More preferably, the microphone 75 is positioned such that it contacts a portion of the user's head 18 near a bone, such that vibrations generated from the user's voice and traveling through the bone, are conducted to the microphone. In another alternative, the microphone 75 can be configured to be inserted into the meatus 24 (FIG. 2) of the ear canal of the user. Thus, in this modification, the microphone 75 can be substituted for one of the speakers 14, 16. Alternatively, an ear-canal type bone conduction microphone can be combined with a speaker so as to provide two-way communication with the user through a single ear canal. Jannard ’526 at 25:12–21 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the audio device 10C is advantageously adapted to support any of a variety of portable electronic circuitry or devices which have previously been difficult to incorporate into conventional headsets due to bulk, weight or other considerations. For example, but without limitation, the electronics are digital or other storage devices and retrieval circuitry such as for retrieving music or other information from MP3 format memory or other memory devices. Jannard ’526 at 27:36–41 The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way

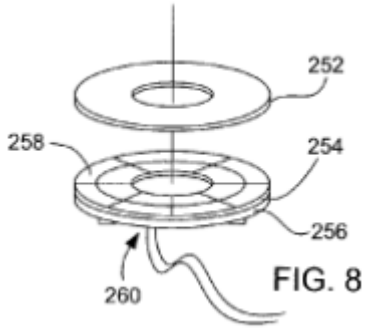
Claims	Prior Art
	communication. For example, the transceiver 114 can be configured to receive a signal from another transmitter and provide audio output to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. Jannard '526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. Rader '474 at 8:7–16 Accessories 10 to the mobile phone may be used in conjunction with the mobile phone and any of the aforementioned subsystems. Accessories 10 may include, for example, monaural or binaural headphones worn by the user. The binaural headphones would allow for gathering of a hearing profile for each ear, which may become more prevalent as mobile phones become stereo audio playback devices. The accessories 10 could communicate with the mobile phone either through direct wire link or wireless link, as shown in FIG. 3.

Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[17d] wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals.	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals." *See* above at claim [1f]. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
18. A head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output signals are wirelessly received from a mobile communication device associated with the user.	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output signals are wirelessly received from a mobile communication device associated with the user." *See* above at claim 2. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[19a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a touch sensitive surface at least partially contained within the head-worn personal electronic apparatus and configured to sense a touch input associated with the user,	DHA systems, alone or in view of Bodley or Hollemans or Seshadri '839, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a touch sensitive surface at least partially contained within the head-worn personal electronic apparatus and configured to sense a touch input associated with the user." *See* above at claim [3a]. To the extent that DHA systems, alone or in view of Bodley or Hollemans or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

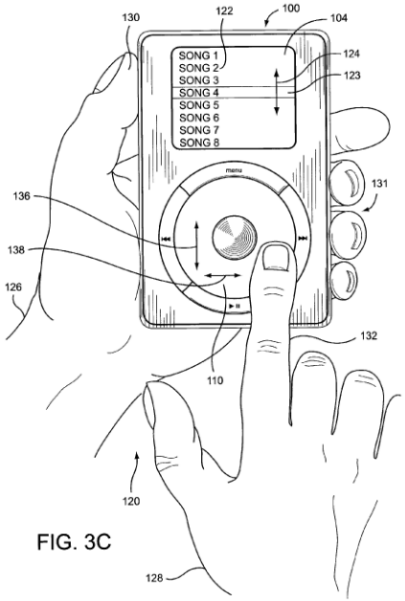
Claims	Prior Art
[19b] wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus.	DHA systems, alone or in view of Bodley or Hollemans '490 or Seshadri '839, expressly or inherently disclose "wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus." *See* above at claim [3b]. A POSITA would have been motivated to combine the teachings of the DHA system with Bodley or Hollemans or Seshadri to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim [19a]. To the extent that DHA systems, alone or in view of Bodley or Hollemans or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
21. A head-worn personal electronic apparatus recited in claim 19, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user.	DHA systems, alone or in view of Jannard, Bodley, or Hollemans, expressly or inherently disclose "[a] head-worn personal electronic apparatus recited in claim 19, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user." *See* above at claim 5. A POSITA would have been motivated to combine the teachings of the DHA system with Jannard, Bodley, or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim [19a]. To the extent that DHA systems, alone or in view of Jannard, Bodley, or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
22. A head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to distinguish at least a direction of the user input with respect to the touch sensitive surface.	DHA systems, alone or in view of Zadesky or Hollemans, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to distinguish at least a direction of the user input with respect to the touch sensitive surface." Zadesky '230 at 2:16–59 With regards to touch pads, mice and track balls, a Cartesian coordinate system is used to monitor the position of the finger, mouse and ball, respectively, as they are moved. The Cartesian coordinate system is generally defined as a two dimensional coordinate system (x, y) in which the coordinates of a point (e.g., position of finger, mouse or ball) are its distances from two intersecting, often perpendicular straight lines,

Claims	Prior Art
	the distance from each being measured along a straight line parallel to each other. For example, the x, y positions of the mouse, ball and finger may be monitored. The x, y positions are then used to correspondingly locate and move the input pointer on the display screen. To elaborate further, touch pads generally include one or more sensors for detecting the proximity of the finger thereto. By way of example, the sensors may be based on resistive sensing, surface acoustic wave sensing, pressure sensing (e.g., strain gauge), optical sensing, capacitive sensing and the like. The sensors are generally dispersed about the touch pad with each sensor representing an x, y position. In most cases, the sensors are arranged in a grid of columns and rows. Distinct x and y position signals, which control the x, y movement of a pointer device on the display screen, are thus generated when a finger is moved across the grid of sensors within the touch pad. For brevity sake, the remaining discussion will be held to the discussion of capacitive sensing technologies. It should be noted, however, that the other technologies have similar features. Capacitive sensing touch pads generally contain several layers of material. For example, the touch pad may include a protective shield, one or more electrode layers and a circuit board. The protective shield typically covers the electrode layer(s), and the electrode layer(s) is generally disposed on a front side of the circuit board. As is generally well known, the protective shield is the part of the touch pad that is touched by the user to implement cursor movements on a display screen. The electrode layer(s), on the other hand, is used to interpret the x, y position of the user's finger when the user's finger is resting or moving on the protective shield. The electrode layer (s) typically consists of a plurality of electrodes that are positioned in columns and rows so as to form a grid array. The columns and rows are generally based on the Cartesian coordinate system and thus the rows and columns correspond to the x and y directions. Zadesky ’230 at 12:31–41 The capacitance in each of the electrodes 258 is measured by control circuitry 260 located on the backside of the circuit board 256. By detecting changes in capacitance at each of the electrodes 258, the control circuitry 260 can determine the angular location, direction, speed and acceleration of the finger as it is moved across the touch pad 250. The control circuitry 260 can also report this information in a form that can be used by a computing device. By way of example, the control circuitry may include an ASIC (application specific integrated circuit). Hollemans ’490 at 8:19–22 In a third way, the touch-sensitive area is used for picking up a gesture made on the area with e.g. a stylus or a finger. The remote control may be extended for this purpose with means for detecting a gesture, e.g. a laser beetle, a small camera, an IR movement sensor, or a cross-capacitance sensor.

Claims	Prior Art
	Hollemans ’490 at 8:30–33 The gesture detected may be a user defined gesture for activating a user defined function. The gesture detected may be used for user identification by matching with initials or an autograph. The gesture may extend into two dimensions, e.g. a planar gesture, or it may extend into three dimensions. A POSITA would have been motivated to combine the teachings of the DHA system with Zadesky or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims [1a] and [19a]. Ax explained above in reference to claim 19, a POSITA would have been motivated to combine the DHA system with the teachings of Bodley ’433 to provide the earpiece with a touch sensitive surface that can be used for controlling functions and settings of the earpiece. Moreover, a POSITA would be motivated to use Zadesky ’230’s capacitive sensing touch pad in the modified DHA system and would have been motivated to look to other references to supply these missing details of implementation. A POSITA would have been motivated to look to Zadesky ’230 or Hollemans at least because it is in the same general field of endeavor. As discussed above, each disclose head-worn devices for providing audio output to a user. Zadesky ’230 discloses a media player that is “dedicated to processing media such as audio” and “allow[s] a user to listen to music.” Zadesky ’230 is therefore from the same general field of endeavor as the Phonak system and Bodley ’433—providing audio output to a user. Thus, a POSITA would have been motivated to combine the teachings of these references with a reasonable expectation of success. To the extent that DHA systems, alone or in view of Zadesky or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
23. A head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to detect at least a sliding of a finger action with respect to the touch sensitive surface.	DHA systems, alone or in view of Zadesky or Hollemans, expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to detect at least a sliding of a finger action with respect to the touch sensitive surface.”

Claims	Prior Art
	Zadesky ’230 at FIG. 8  FIG. 8 Zadesky ’230 at 11:50–12:20 FIG. 8 is a partially broken away perspective view of an annular capacitive touch pad 250, in accordance with one embodiment of the present invention. By way of example, the annular capacitive touch pad 250 may correspond to the touch pad of FIG. 2. The annular capacitive touch pad 250 is arranged to detect changes in capacitance as the user swirls an object such as a finger around the touch pad 250. The annular capacitive touch pad 250 is also arranged to detect changes in capacitance as the user moves their finger radially across the touch pad 250. The annular capacitive touch pad 250 is formed from various layers including at least a label layer 252, an electrode layer 254 and a circuit board 256. The label layer 252 is disposed over the electrode layer 254 and the electrode layer 254 is disposed over the circuit board 256. At least the label 252 and electrode layer 254 are annular such that they are defined by concentric circles, i.e., they have an inner perimeter and an outer perimeter. The circuit board 256 is generally a circular piece having an outer perimeter that coincides with the outer perimeter of the label 252 and electrode layer 254. It should be noted, however, that in some cases the circuit board 256 may be annular or the label 252 and electrode layer 254 may be circular. The label layer 252 serves to protect the underlayers and to provide a surface for allowing a finger to slide thereon. The surface is generally smooth so that the finger does not stick to it when moved. The label layer 252 also provides an insulating layer between the finger and the electrode layer 254. The electrode layer 254 includes a plurality of spatially distinct electrodes 258 that have positions based on the polar coordinate system. For instance, the electrodes 258 are positioned both angularly and radially on the circuit board 256 such that each of the electrodes 258 defines a distinct angular and radial position thereon. Any suitable number of electrodes 258 may be used. In most cases, it would be desirable to increase the number of electrodes 258 so as to provide higher resolution, i.e., more information can be used for things such as acceleration.

Claims	Prior Art
	Hollemans ’490 at 8:19–22 In a third way, the touch-sensitive area is used for picking up a gesture made on the area with e.g. a stylus or a finger. The remote control may be extended for this purpose with means for detecting a gesture, e.g. a laser beetle, a small camera, an IR movement sensor, or a cross-capacitance sensor. Hollemans ’490 at 8:30–33 The gesture detected may be a user defined gesture for activating a user defined function. The gesture detected may be used for user identification by matching with initials or an autograph. The gesture may extend into two dimensions, e.g. a planar gesture, or it may extend into three dimensions. A POSITA would have been motivated to combine the teachings of the DHA system and Zadesky ’230 or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 19. To the extent that DHA systems, alone or in view of Zadesky or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
24. A head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to facilitate determination at least whether the user input pertains to a forward sliding action or a backward sliding action with respect to the touch sensitive surface.	DHA systems, alone or in view of Zadesky or Hollemans, expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus is configured to facilitate determination at least whether the user input pertains to a forward sliding action or a backward sliding action with respect to the touch sensitive surface.”

Claims	Prior Art
	Zadesky ’230 at FIG. 3C  FIG. 3C Zadesky ’230 at 7:28–44 The touch pad generally consists of a touchable outer surface 111 for receiving a finger for manipulation on the touch pad 110. Although not shown in FIG. 2, beneath the touchable outer surface 111 is a sensor arrangement. The sensor arrangement includes a plurality of sensors that are configured to activate as the finger passes over them. In the simplest case, an electrical signal is produced each time the finger passes a sensor. The number of signals in a given time frame may indicate location, direction, speed and acceleration of the finger on the touch pad, i.e., the more signals, the more the user moved his or her finger. In most cases, the signals are monitored by an electronic interface that converts the number, combination and frequency of the signals into location, direction, speed and acceleration information. This information may then be used by the media player 100 to perform the desired control function on the display screen 104. Zadesky ’230 at 8:52–67 FIGS. 3A-3C show the media player 100 of FIG. 2 being used by a user 120, in accordance with different embodiments of the invention. In all of these embodiments, the user 120 is linearly scrolling 104 (as shown by arrow 124) through a list of songs 122 displayed on the display screen via a slider bar 123. As

Claims	Prior Art
	shown, the media device 100 is comfortably held by one hand 126 while being comfortably addressed by the other hand 128. This configuration generally allows the user 120 to easily actuate the touch pad 110 with one or more fingers. For example, the thumb 130 and rightmost fingers 131 (or leftmost fingers if left handed) of the first hand 126 are used to grip the sides of the media player 100 while a finger 132 of the opposite hand 128 is used to actuate the touch pad 110. As shown, the entire top surface of the touch pad 110 is accessible to the user's finger 130. Zadesky ’230 at 9:16–21 Referring to FIG. 3C, and in accordance with one embodiment of the invention, the user 120 can slide his or her finger 132 substantially tangentially from all sides of the touch pad 110. For example, the touch pad 110 may be actuated forwards and backwards as shown by arrows 136 and side to side by arrows 138. Zadesky ’230 at 11:50–12:20 FIG. 8 is a partially broken away perspective view of an annular capacitive touch pad 250, in accordance with one embodiment of the present invention. By way of example, the annular capacitive touch pad 250 may correspond to the touch pad of FIG. 2. The annular capacitive touch pad 250 is arranged to detect changes in capacitance as the user swirls an object such as a finger around the touch pad 250. The annular capacitive touch pad 250 is also arranged to detect changes in capacitance as the user moves their finger radially across the touch pad 250. The annular capacitive touch pad 250 is formed from various layers including at least a label layer 252, an electrode layer 254 and a circuit board 256. The label layer 252 is disposed over the electrode layer 254 and the electrode layer 254 is disposed over the circuit board 256. At least the label 252 and electrode layer 254 are annular such that they are defined by concentric circles, i.e., they have an inner perimeter and an outer perimeter. The circuit board 256 is generally a circular piece having an outer perimeter that coincides with the outer perimeter of the label 252 and electrode layer 254. It should be noted, however, that in some cases the circuit board 256 may be annular or the label 252 and electrode layer 254 may be circular. The label layer 252 serves to protect the underlayers and to provide a surface for allowing a finger to slide thereon. The surface is generally smooth so that the finger does not stick to it when moved. The label layer 252 also provides an insulating layer between the finger and the electrode layer 254. The electrode layer 254 includes a plurality of spatially distinct electrodes 258 that have positions based on the polar coordinate system. For instance, the electrodes 258 are positioned both angularly and radially on the circuit board 256 such that each of the electrodes 258 defines a distinct angular and radial position thereon. Any suitable number of electrodes 258 may be used. In most cases, it would be desirable to

Claims	Prior Art
	increase the number of electrodes 258 so as to provide higher resolution, i.e., more information can be used for things such as acceleration. Hollemans '490 at 8:19–22 In a third way, the touch-sensitive area is used for picking up a gesture made on the area with e.g. a stylus or a finger. The remote control may be extended for this purpose with means for detecting a gesture, e.g. a laser beetle, a small camera, an IR movement sensor, or a cross-capacitance sensor. Hollemans '490 at 8:30–33 The gesture detected may be a user defined gesture for activating a user defined function. The gesture detected may be used for user identification by matching with initials or an autograph. The gesture may extend into two dimensions, e.g. a planar gesture, or it may extend into three dimensions. A POSITA would have been motivated to combine the teachings of the DHA system and Zadesky '230 or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 19. To the extent that DHA systems, alone or in view of Zadesky or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
25. A head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus comprises: a sensor at least partially contained within the head-worn personal electronic structure and configured to sense data indicative of whether the head-worn personal electronic apparatus is being worn by the user.	DHA systems, alone or in view of Abreu or Luo '446 or Irvin '332, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 19, wherein the head-worn personal electronic apparatus comprises: a sensor at least partially contained within the head-worn personal electronic structure and configured to sense data indicative of whether the head-worn personal electronic apparatus is being worn by the user." *See* above at claim [13a], [13b], and [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 13. To the extent that DHA systems, alone or in view of Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with

Claims	Prior Art
	the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
26. A head-worn personal electronic apparatus as recited in claim 25, wherein the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the sensed data from the sensor.	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 25, wherein the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the sensed data from the sensor." *See* above at claim [14b]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 13. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
27. A head-worn personal electronic apparatus as recited in claim 26, wherein the operation of the head-worn personal electronic apparatus being altered concerns control of the audio output being provided to the user.	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 26, wherein the operation of the head-worn personal electronic apparatus being altered concerns control of the audio output being provided to the user." Abreu's eyeglasses can "generate an identifiable signal" based on an abrupt pressure drop detected by pressure sensing means in the eyeglasses. Abreu '960 at 15:63-65. Abreu further discloses that signals produced by the eyeglasses can be reported by "audio, visual or tactile means." Abreu '960 at 19:25-26. Therefore, a POSITA would understand that that the operation being altered based on the sensed data from the pressure sensing means (the generation of an identifiable signal) concerns control of audio output being provided to a user in that it relates to whether or not an audio signal is output. Similarly, Irvin discloses that entering an "inactive state" upon determining that the device has been removed from the user's body, and that the device is "non-functional" in the low-power state. Irvin '332 at 8:11–22. A POSITA would understand that that the operation being altered based on the sensed data from Irvin's sensor includes terminating audio output provided to a user. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 13.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
28. A head-worn personal electronic apparatus as recited in claim 26, wherein the operation of the head-worn personal electronic apparatus being altered concerns operation of the head-worn personal electronic apparatus in a lower power operation.	DHA systems, alone or in view of Anderson or Jannard or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 26, wherein the operation of the head-worn personal electronic apparatus being altered concerns operation of the head-worn personal electronic apparatus in a lower power operation." Phonak MicroLink MLxS Product Information at Page 1  • Energy-saving, intelligent standby mode Anderson '783 at 21:31–34 Location of misplaced system components: Modern CIC hearing aid earpieces 10 of the type possibly used in this invention may be smaller than the head of a cotton swab, and are easily misplaced (as are small RPUs 16). Anderson '783 at 21:34–40 If the two-way wireless link between an earpiece and RPU 17 is interrupted for a certain period of time, as occurs when an RPU is misplaced, the earpiece automatically simulates stand-alone CIC hearing aid operation and the RPU, detecting a link disruption as described earlier, conserves power by reducing its number of attempts to communicate with the earpiece. Jannard '526 at 21:22–34 In one implementation, the transceiver 114 can operate at least two power levels: a lower power level that covers a range of about ten yards and a higher power level. The higher level covers a range of about one hundred yards, can function even in very noisy radio environments, and can be audible under severe conditions. The transceiver 114 can advantageously limit its output with reference to system requirements. For example, without limitation, if the source electronics B is only a short distance from audio device 10C, the transceiver 114 modifies its signal to be suitable for the distance. In another implementation, the transceiver 114 can switch to a low-power mode when traffic volume becomes low or stops.

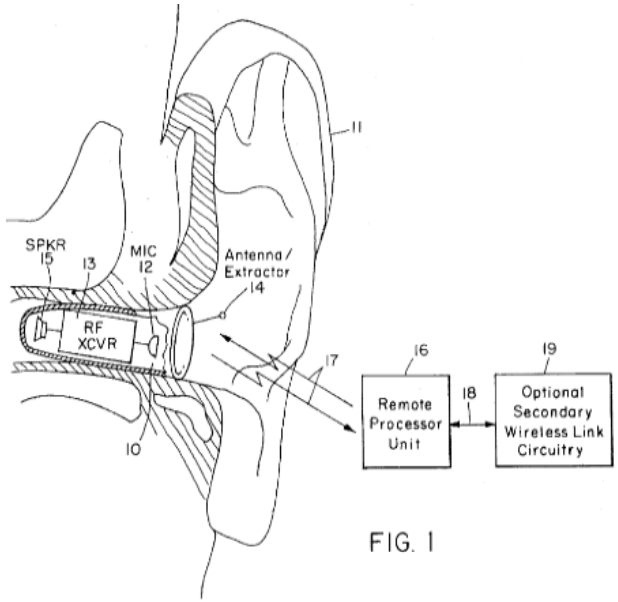
Claims	Prior Art
	Jannard ’526 at 29:2–8 As is schematically illustrated in FIG. 19, when the signal detector 236 detects a 49 MHz signal, which corresponds to a state in which no audio signal is being transmitted from the MP3 player 206, the carrier detector 246 causes the transceiver 114 to enter a sleep mode, schematically illustrated by the operation block 248. Irvin ’332 at 2:20–34 The wireless headset serves as a communication accessory to a base unit, such as a cellular telephone. When not in use, the wireless headset reverts to an inactive state and draws essentially no power. A power control circuit imparts automatic enabling capability to the wireless headset. Preferably, the power control circuit employs a sensor responsive to a natural stimulus associated with a person preparing to use the wireless headset. The wireless headset may sense motion, physical orientation, or user proximity. Sensing techniques include but are not limited to micro-electro-mechanical-systems (MEMS), mercury switches, attitude switches, and IR or acoustic proximity sensors. In some embodiments, the wireless headset transitions from the inactive state to an active state in response to a user picking it up or mounting the wireless headset upon their head. Irvin ’332 at 6:3–33 FIG. 4C illustrates the power control circuit 210 coupled with a contact switch functioning as the sensor 220. The contact switch includes a first and second contact connected to the power control circuit 210, and connecting member 222 actuated when the user places the wireless headset 100 on their body. When actuated, the connecting member 222 places the first and second contacts of sensor 220 in electrical connection. Ideally, the sensor 220, when configured as a contact sensor, is placed in the earpiece 130, in the retaining member 140, or in the retaining pad 150 shown in FIG. 1B. Preferably, when sensor 220 is configured as a contact-based proximity sensor, the contacting element of sensor 220 is sealed in a manner that minimizes contaminant ingress. In some embodiments, the contact sensor may be actuated by the user resiliently expanding the retaining member 140 or other element of the wireless headset 100 for mounting upon their body. FIG. 4D illustrates the power control circuit 210 coupled with a proximity sensor 220 not based on mechanical actuation. The proximity sensor 220 may be an active or passive IR sensor, may be a resistive or other type of thermal sensor, or may be an acoustic sensor. As a proximity sensor, sensor 220 provides indication that a user has placed the wireless headset 100 upon their body. As such, any of the foregoing or other presence sensing techniques may be used to implement sensor 220. Preferably, when configured as a proximity sensor, sensor 220 requires minimal power and provides a sensor output signal based on

Claims	Prior Art
	proximity to the user's body. Depending upon the sensing technology chosen, the control circuit 210 may be required to periodically stimulate sensor 220 to generate a detection response. Irvin '332 at 6:34–48 FIG. 4E illustrates the power control circuit 210 coupled to an alternate type of attitude sensor 220. In this embodiment, the attitude sensor 220 is a mechanical sensor that includes a contacting member 224 responsive to a physical orientation of the sensor 220. In one or more physical orientations, the contacting member 224 does not electrically connect the first and second contacts of attitude sensor 220. In one or more other physical orientations, the contacting member places the first and second contacts of attitude sensor 220 in electrical connection. The power control circuit 210 is configured to be responsive to this electrical connection. Contacting member 224 may be resiliently biased away from one or both contacts to ensure an open electrical connection between contacts when the headset is not oriented in a position indicative of use. Irvin '332 at 8:11–22 FIGS. 6 and 7 illustrate operation of the wireless headset 100. FIG. 6 is a state diagram for one exemplary embodiment of the present invention. When the wireless headset 100 is not in use, it automatically enters an inactive state (610). As explained, this inactive state may constitute disconnecting the communications circuit 230 from the power source 240, or may simply represent placing the communications circuit 230 in a non-functional, low-power condition. Absent user stimulus, such as motion of the wireless headset 100 or placement of the wireless headset 100 upon the user's body, the wireless headset 100 remains in the inactive state (610). A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. A POSITA would have been further motivated to combine the teachings of DHA system and Anderson or Jannard or Irvin, with a reasonable expectation of success, for the reasons described above with respect to claim 13. It would have been obvious to a POSITA to modify the DHA system to include Irvin's teachings to operate the audio device in a lower power mode if the audio device is not being worn because a POSITA would recognize this as another instance of the audio device not being needed, and an efficient way to conserve battery without interfering with a user's use of the device. A POSITA would have been motivated to include this feature in the proposed combination with a reasonable expectation of success, for the reasons described above with respect to claims 13, 14, and 25. To the extent that DHA systems, alone or in view of Anderson or Jannard or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to

Claims	Prior Art
	combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[29a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user." Phonak – Perseo Brochure at Page 23 

Claims	Prior Art
	Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user. Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed

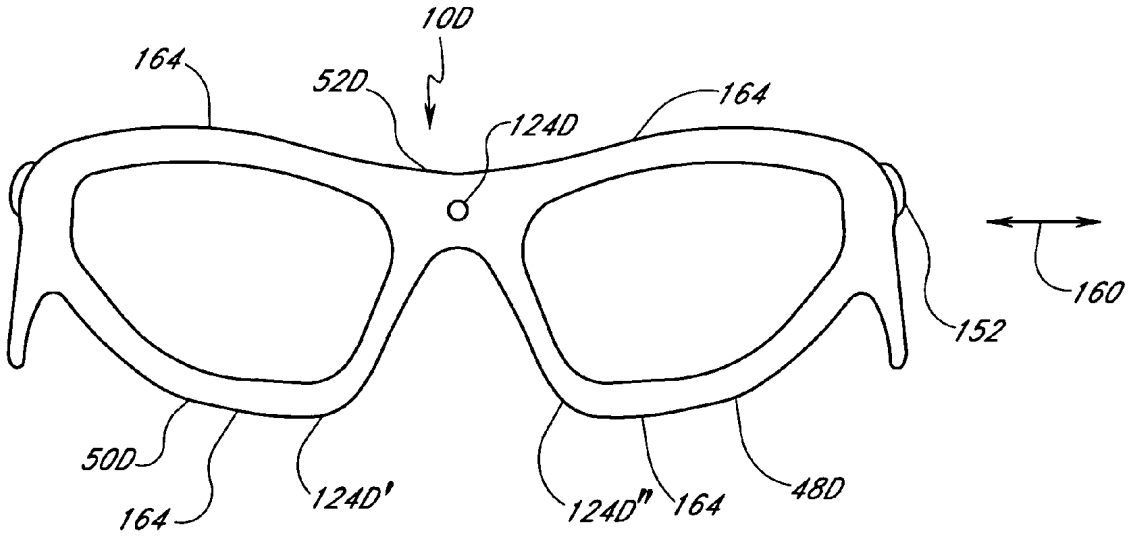
Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode). Phonak – Collections 2002 Product List at Page 7  Adaptive digital AudioZoom (dAZ) Fine-scale Noise Canceler (FNC) Active and adaptive separation of speech from noise The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically. Fully automatic, fully adaptive With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR. Twenty-band noise control Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal. Precise and selective All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected. Intelligent The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved. 7

Claims	Prior Art
	Anderson ’783 at FIG. 1  FIG. 1 Anderson ’783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the

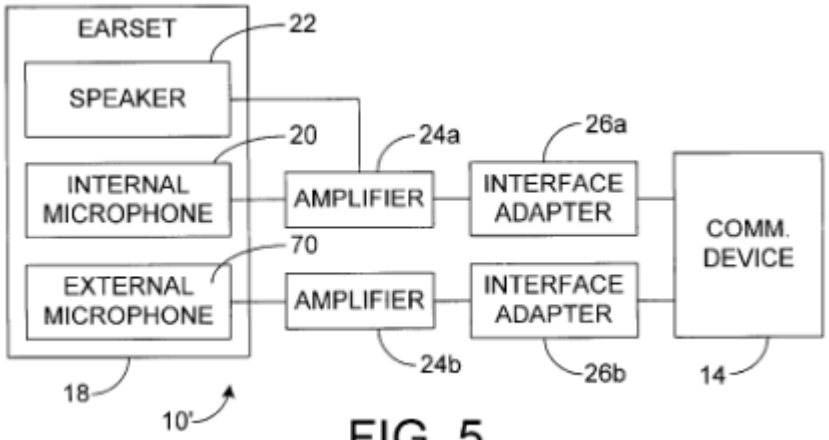
Claims	Prior Art
	RPU case 70 of FIG. 7. Anderson ’783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an

Claims	Prior Art
	earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art

Claims	Prior Art
	auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 23:36–42 Binaural, directional noise-canceling hearing aid with hearing protection: Two earpieces (one in each ear) and an RPU can be used in a binaural, directional noise-canceling hearing aid system with hearing protection features. Such a system provides some measure of protection for the residual hearing capability of a hearing-impaired user simultaneous with the hearing aid function. Anderson ’783 at 25:29–30 The verbal commands are received as audio signals by the earpiece 10 and transmitted to the RPU 16. Anderson ’783 at 26:54–65 Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Anderson ’783 at 26:65–67 Note that such binaural signal processing techniques can also provide directivity and background noise

Claims	Prior Art
	cancellation for users Anderson ’783 at 27:1–3 having total hearing loss in one ear and partial loss in the other ear, as discussed earlier in connection with binaural hearing aid system operation. Jannard ’526 at FIG. 11  FIG. 11 Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted

Claims	Prior Art
	telecommunication signal. Jannard ’526 at 2:26–34 In accordance with another embodiment of at least one of the inventions disclosed herein, an eyeglass frame comprises a support for supporting at least one lens in the path of a wearer's field of view, a first ear stem attached to the support, a second ear stem attached to the support, and at least one microphone supported by at least one of the support, first ear stem, and second ear stem. The microphone is advantageously arranged to face towards the head of a wearer of the eyeglass frame. Jannard ’526 at 22:1–9 In the audio device 10D, the microphone 124D can be disposed in the frame 42D. In particular, the microphone 124D can be disposed in the bridge portion 52D. Alternatively, the microphone 124D can be disposed along a lower edge of the right orbital 50D, this position being identified by the reference numeral 124D′. Further, the microphone could be positioned in a lower edge of the left orbital 48D, this position being identified by the reference numeral 124D″. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different

Claims	Prior Art
	directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or

Claims	Prior Art
	wireless, telephone from an RF transceiver of the telephone. Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[29b] a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose “a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user.”

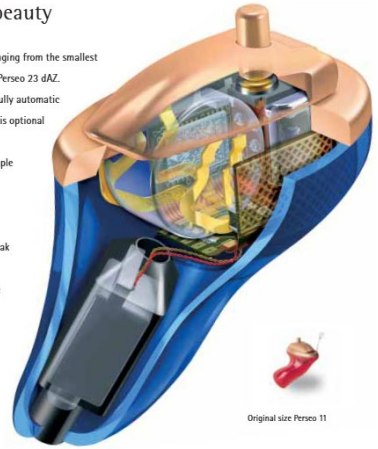
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.



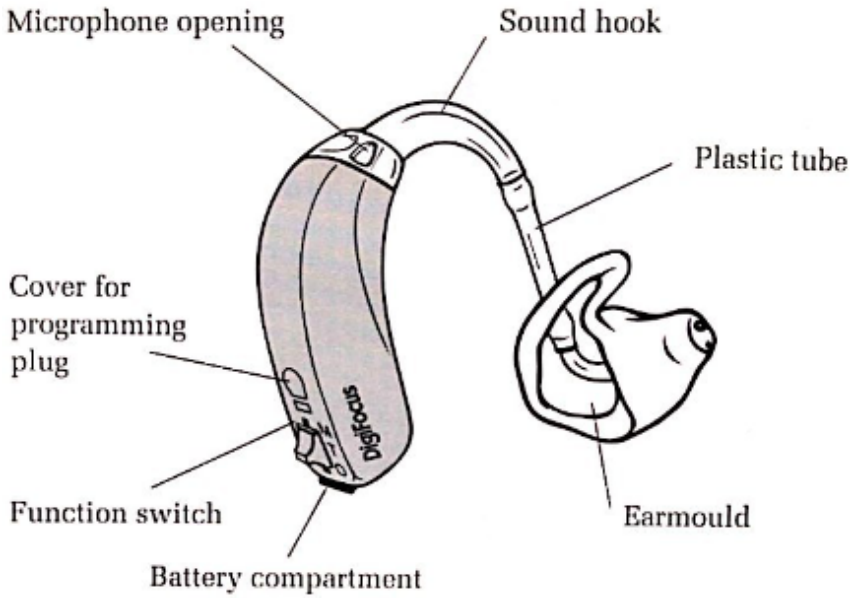
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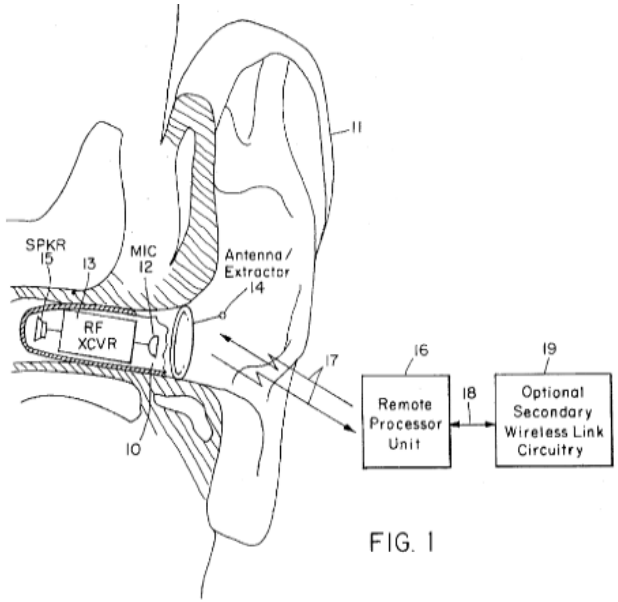
IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
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Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed

Claims	Prior Art
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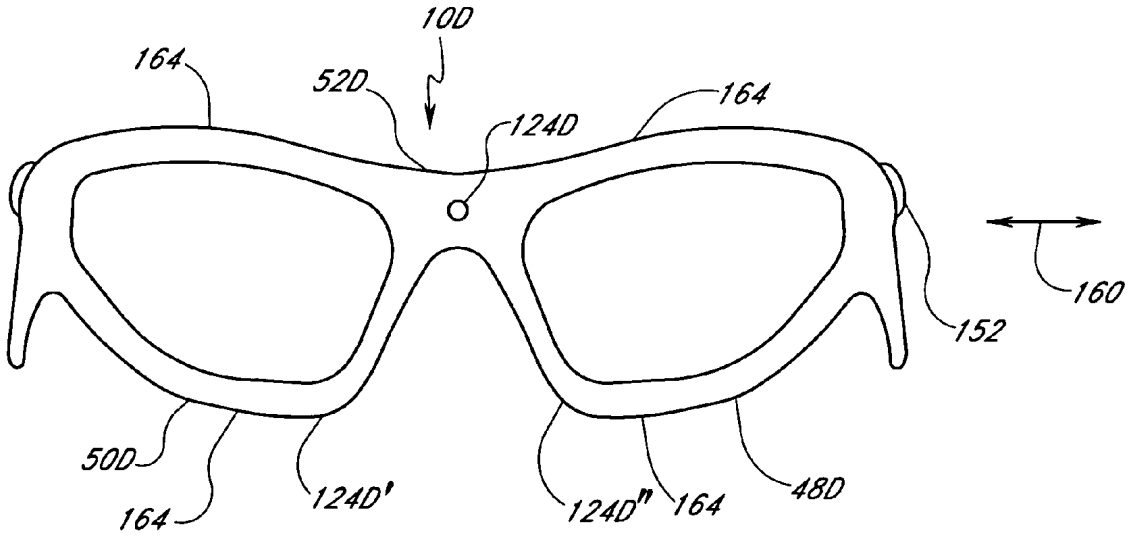
Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4 

Claims	Prior Art
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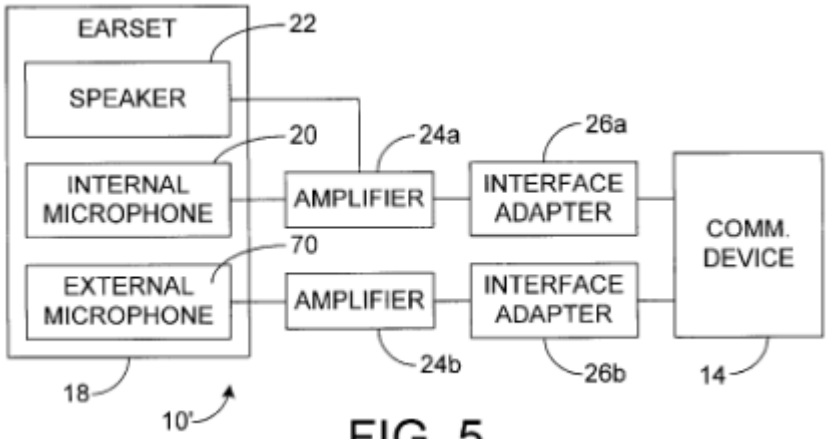
Claims	Prior Art
	RPU case 70 of FIG. 7. Anderson ’783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an

Claims	Prior Art
	earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art

Claims	Prior Art
	auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 23:36–42 Binaural, directional noise-canceling hearing aid with hearing protection: Two earpieces (one in each ear) and an RPU can be used in a binaural, directional noise-canceling hearing aid system with hearing protection features. Such a system provides some measure of protection for the residual hearing capability of a hearing-impaired user simultaneous with the hearing aid function. Anderson ’783 at 25:29–30 The verbal commands are received as audio signals by the earpiece 10 and transmitted to the RPU 16. Anderson ’783 at 26:54–65 Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Anderson ’783 at 26:65–67 Note that such binaural signal processing techniques can also provide directivity and background noise

Claims	Prior Art
	cancellation for users Anderson ’783 at 27:1–3 having total hearing loss in one ear and partial loss in the other ear, as discussed earlier in connection with binaural hearing aid system operation. Jannard ’526 at FIG. 11  FIG. 11 Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted

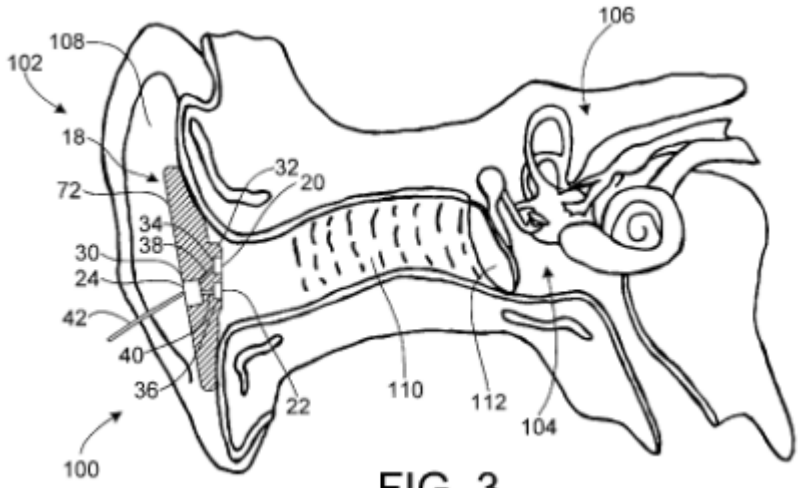
Claims	Prior Art
	telecommunication signal. Jannard ’526 at 22:1–9 In the audio device 10D, the microphone 124D can be disposed in the frame 42D. In particular, the microphone 124D can be disposed in the bridge portion 52D. Alternatively, the microphone 124D can be disposed along a lower edge of the right orbital 50D, this position being identified by the reference numeral 124D′. Further, the microphone could be positioned in a lower edge of the left orbital 48D, this position being identified by the reference numeral 124D″. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes.

Claims	Prior Art
	Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[30a] A head-worn personal electronic apparatus as recited in claim 29, wherein the electronic circuitry is configured to: determine an ambient sound level; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 29, wherein the electronic circuitry is configured to: determine an ambient sound level." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[30b] modify audio output to the at least one speaker dependent at least in part on the ambient sound level.	DHA systems, alone or in view of Jannard, Nemirovski, or Victorian, expressly or inherently disclose "modify audio output to the at least one speaker dependent at least in part on the ambient sound level." *See* above at claim [12e]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Nemirovski or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Jannard, Nemirovski, or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[31a] A head-worn personal electronic apparatus as recited in claim 29, wherein the electronic circuitry is configured to: produce an ambient audio output signal	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 29, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input." *See* above at claim [9a].

Claims	Prior Art
based on at least the ambient sound input; and	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[31b] provide at least the ambient audio output signal to the at least one speaker.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "provide at least the ambient audio output signal to the at least one speaker." *See* above at claim [9b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone of in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[32a] A head-worn personal electronic apparatus as recited in claim 31, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input,	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 31, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[32b] wherein the ambient audio output signal is dependent on the ambient noise level.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the ambient audio output signal is dependent on the ambient noise level." *See* above at claim [10b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[33a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus." *See* above at claim [29a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[33b] a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being at least partially in the head-worn personal electronic apparatus,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski expressly or inherently disclose "a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being at least partially in the head-worn personal electronic apparatus." *See* above at claim [29b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[33c] wherein the second microphone is positioned to be further away from mouth of the user than the first microphone when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn.	DHA systems, alone or in view of Nemirovski, expressly or inherently disclose "wherein the second microphone is positioned to be further away from mouth of the user than the first microphone when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn." Nemirovski '379 at FIG. 3  FIG. 3 Nemirovski '379 at 2:54–56 FIG. 3 is a cross-sectional view of an ear microphone assembly, or earset, disposed with respect to the ear of a person, and the ear also shown in cross-section; Nemirovski '379 at 6:36–41 The microphone 20 is disposed on an interior portion 32 of the housing 30. Accordingly, the interior portion 32 is sized and shaped to position the microphone 20 with respect to the ear 100 and/or ear canal 110 as desired. In one embodiment, the interior portion 32 positions the microphone 20 and/or the speaker 22 in the concha 122.

Claims	Prior Art
	Nemirovski ’379 at 9:19–25 With additional reference to FIG. 5, in another embodiment of the present invention, the earset 18 can include an external microphone 70 in addition to the microphone 20 directed toward the ear canal of the user (also referred to herein as an internal microphone 20). The external microphone 70 detects air pressure changes, or sounds, originating outside the user's ear. Nemirovski ’379 at 9:33–39 The external microphone 70 is disposed on a exterior portion 72 of the housing 30 and is pointed away from the wearer of the earset 18 assembly. The external microphone 70 can be secured to the housing 30 with adhesive and can be disposed in a recess defined by the exterior portion 72. Similar to the internal microphone 20, the external microphone 70 is small (e.g., less than two mm in size). A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[34a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone; and	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone.”

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

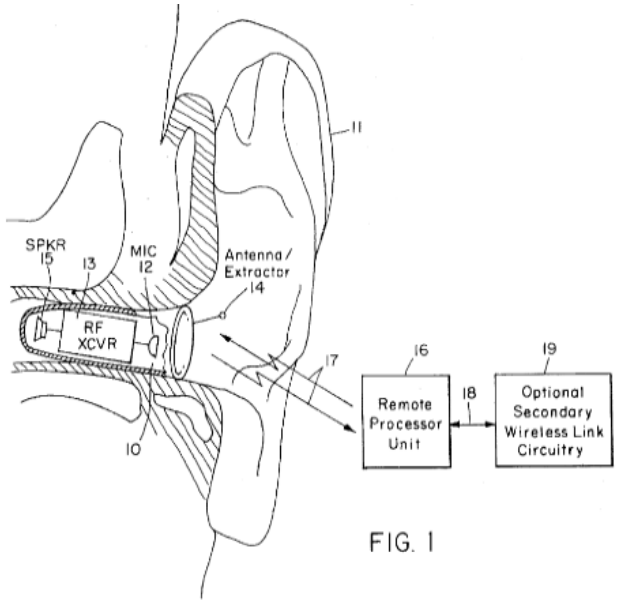
Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

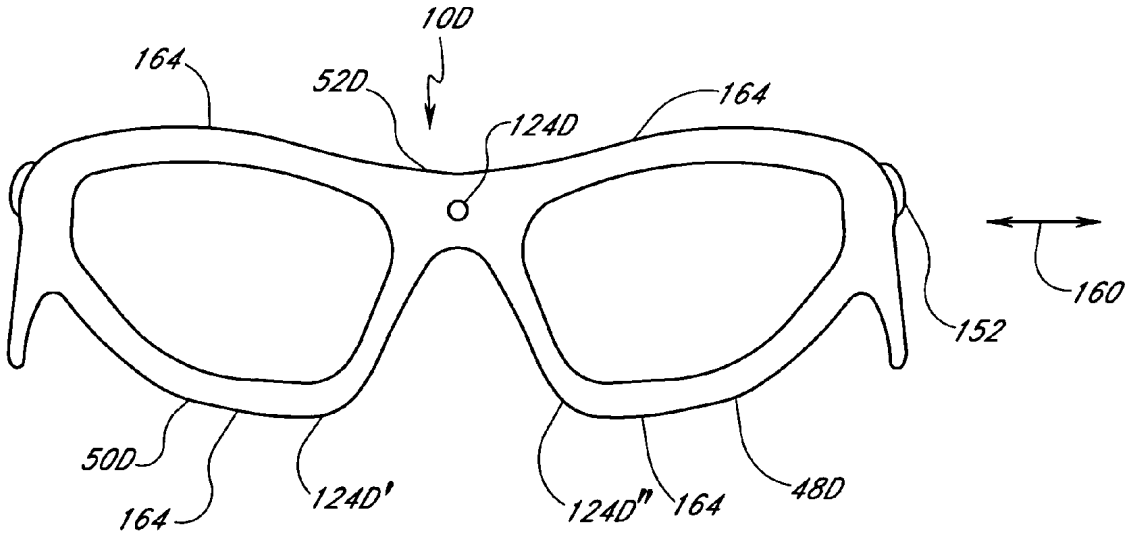
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Claims	Prior Art
	Anderson ’783 at FIG. 1  FIG. 1 Anderson ’783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the

Claims	Prior Art
	RPU case 70 of FIG. 7. Anderson ’783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an

Claims	Prior Art
	earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art

Claims	Prior Art
	auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 23:36–42 Binaural, directional noise-canceling hearing aid with hearing protection: Two earpieces (one in each ear) and an RPU can be used in a binaural, directional noise-canceling hearing aid system with hearing protection features. Such a system provides some measure of protection for the residual hearing capability of a hearing-impaired user simultaneous with the hearing aid function. Anderson ’783 at 25:29–30 The verbal commands are received as audio signals by the earpiece 10 and transmitted to the RPU 16. Anderson ’783 at 26:54–65 Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Anderson ’783 at 26:65–67 Note that such binaural signal processing techniques can also provide directivity and background noise

Claims	Prior Art
	cancellation for users Anderson ’783 at 27:1–3 having total hearing loss in one ear and partial loss in the other ear, as discussed earlier in connection with binaural hearing aid system operation. Jannard ’526 at FIG. 11  FIG. 11 Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted

Claims	Prior Art
	telecommunication signal. Jannard ’526 at 22:1–9 In the audio device 10D, the microphone 124D can be disposed in the frame 42D. In particular, the microphone 124D can be disposed in the bridge portion 52D. Alternatively, the microphone 124D can be disposed along a lower edge of the right orbital 50D, this position being identified by the reference numeral 124D′. Further, the microphone could be positioned in a lower edge of the left orbital 48D, this position being identified by the reference numeral 124D″. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[34b] a second microphone;	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "a second microphone." *See* above at claim [34a]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[34c] wherein the first microphone is configured to pick up at least audio input from at least the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the first microphone is configured to pick up at least audio input from at least the user." *See* above at claim [29a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[34d] wherein the second microphone is configured to pick up at least environmental audio input associated with an environment in the vicinity of the head-worn wearable electronic device, and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the second microphone is configured to pick up at least environmental audio input associated with an environment in the vicinity of the head-worn wearable electronic device."

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

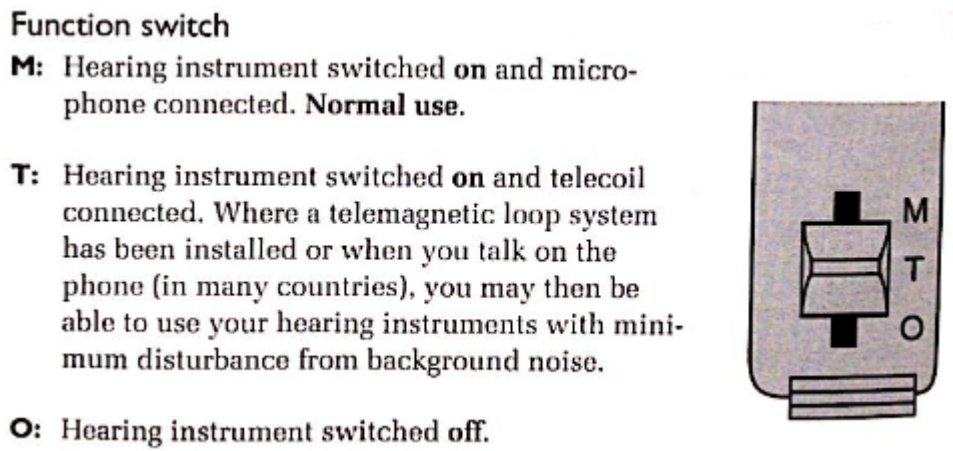
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

7

Oticon – DigiFocus Compact User Instructional Brochure at Page 4

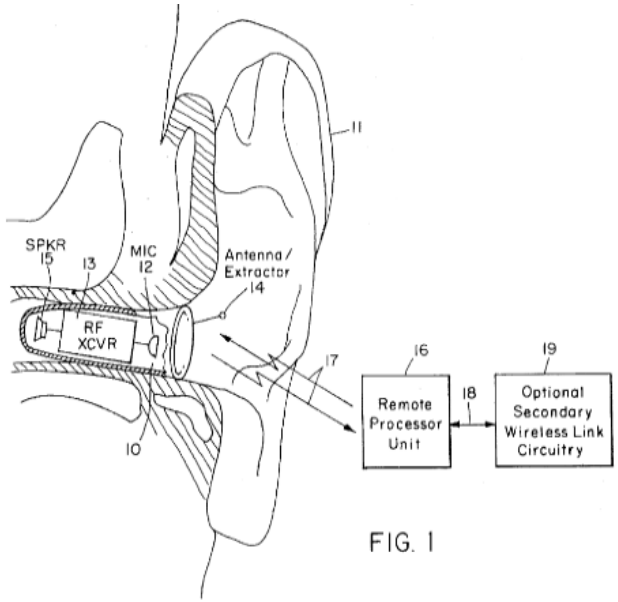


Function switch

M: Hearing instrument switched **on** and microphone connected. **Normal use.**

T: Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise.

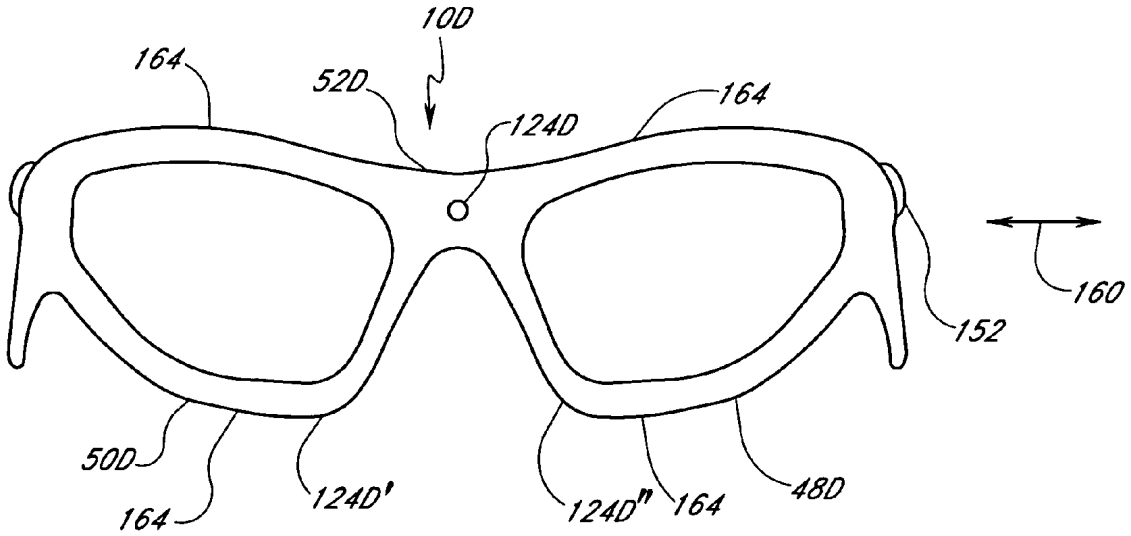
O: Hearing instrument switched **off.**

Claims	Prior Art
	Anderson ’783 at FIG. 1  FIG. 1 Anderson ’783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the

Claims	Prior Art
	RPU case 70 of FIG. 7. Anderson ’783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an

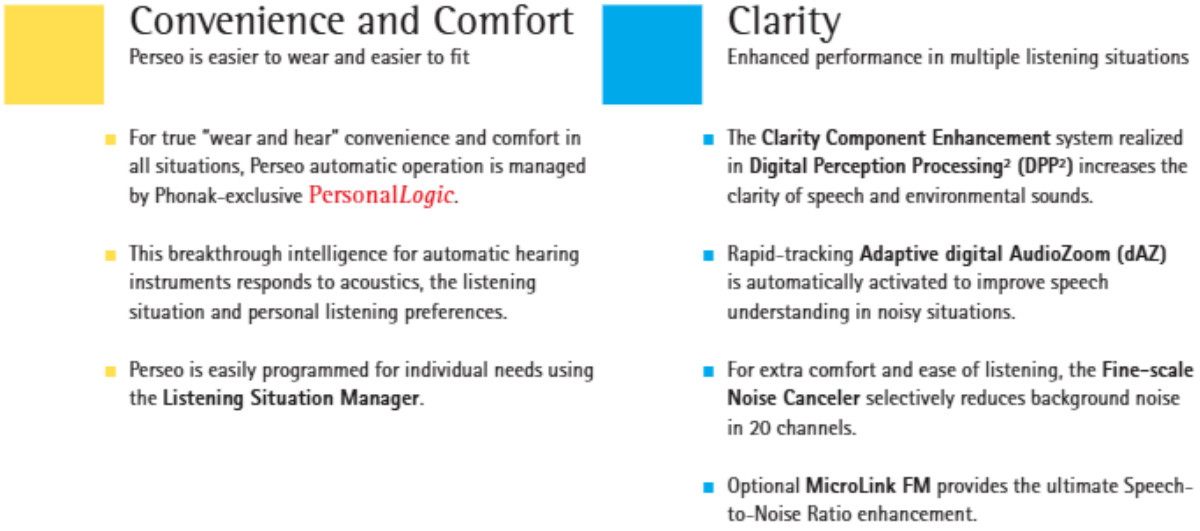
Claims	Prior Art
	earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art

Claims	Prior Art
	auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 23:36–42 Binaural, directional noise-canceling hearing aid with hearing protection: Two earpieces (one in each ear) and an RPU can be used in a binaural, directional noise-canceling hearing aid system with hearing protection features. Such a system provides some measure of protection for the residual hearing capability of a hearing-impaired user simultaneous with the hearing aid function. Anderson ’783 at 25:29–30 The verbal commands are received as audio signals by the earpiece 10 and transmitted to the RPU 16. Anderson ’783 at 26:54–65 Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Anderson ’783 at 26:65–67 Note that such binaural signal processing techniques can also provide directivity and background noise

Claims	Prior Art
	cancellation for users Anderson ’783 at 27:1–3 having total hearing loss in one ear and partial loss in the other ear, as discussed earlier in connection with binaural hearing aid system operation. Jannard ’526 at FIG. 11  FIG. 11 Jannard ’526 at 1:58–2:5 In accordance with one embodiment of at least one of the inventions disclosed herein, a wearable wireless audio interface comprises a support. The support is configured to support at least one lens in a wearer's field of view. The support also comprises a first ear stem and an orbital. A first earphone is supported by the support, directed toward at least one of the wearer's ears, and configured to convert at least one received telecommunication signal into sound. A first electronics device is supported by the support and configured to receive the received telecommunication signal. A microphone is supported by the support and configured to convert the wearer's voice into at least one transmitted telecommunication signal. A second electronics device is supported by the support and configured to transmit the transmitted

Claims	Prior Art
	telecommunication signal. Jannard ’526 at 18:23–26 Further, the audio device 10A″ can include noise cancellation electronics (not shown) configured to filter wind-generated noise from an audio signal transmitted from the microphone 75. Jannard ’526 at 22:1–9 In the audio device 10D, the microphone 124D can be disposed in the frame 42D. In particular, the microphone 124D can be disposed in the bridge portion 52D. Alternatively, the microphone 124D can be disposed along a lower edge of the right orbital 50D, this position being identified by the reference numeral 124D′. Further, the microphone could be positioned in a lower edge of the left orbital 48D, this position being identified by the reference numeral 124D″. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-

Claims	Prior Art
	cancellation purposes. Nemirovski ’379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[34e] wherein the electronic circuitry is configured to process the audio input from at least the user picked-up via the first microphone to remove at least a portion of the environmental audio input associated with the environment picked-up via the second microphone.	DHA systems, alone or in view of Anderson, Jannard or Nemirovski, expressly or inherently disclose “wherein the electronic circuitry is configured to process the audio input from at least the user picked-up via the first microphone to remove at least a portion of the environmental audio input associated with the environment picked-up via the second microphone.”

Claims	Prior Art
	Phonak – Perseo Brochure at Page 7  **Convenience and Comfort** Perseo is easier to wear and easier to fit ■ For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. ■ This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. ■ Perseo is easily programmed for individual needs using the **Listening Situation Manager**. **Clarity** Enhanced performance in multiple listening situations ■ The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. ■ Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. ■ For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. ■ Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement. Phonak – Perseo Brochure at Page 11 ■ With **dual-base automatic** the fitter has direct access to the two separate base-programs. Performance can be matched to each user's needs in quiet situations without disrupting performance in noisy situations and vice versa. The result is less fine-tuning effort than with single-base instruments. With **single-base automatic** time is wasted "juggling" a single set of fitting parameters while attempting to balance performance in multiple listening situations. ■ To further simplify fitting, PersonalLogic is pre-set according to typical needs and is easily re-programmed to individual preferences using the **Listening Situation Manager (LSM)**. Ensuring that an automatic hearing system responds correctly in noisy situations where there is no speech is an important consideration for safety and to eliminate feelings of isolation. The **LSM** provides this functionality with a single mouse click. Matching the system's sensitivity to noise with that of the client's, and controlling the auditory contrast between quiet and noisy settings is equally easy.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 13  Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer. Phonak – Perseo Brochure at p. 14 **Rapid-tracking dAZ** Phonak pioneered the development of directional microphone systems. Not surprisingly, Perseo **dAZ** with new soft switching brings even better performance. Perseo **dAZ** is automatically activated in appropriate situations and immediately reduces noise from beside and behind the wearer. The wearer experiences an immediate reduction of background noise and enhanced speech clarity. Rapid tracking ensures that even fast moving noise sources are controlled.

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear! Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

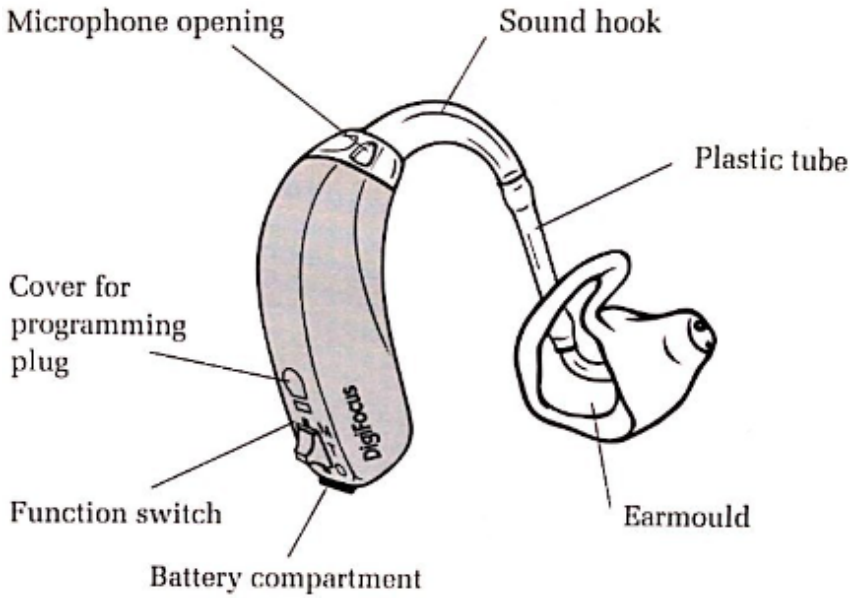
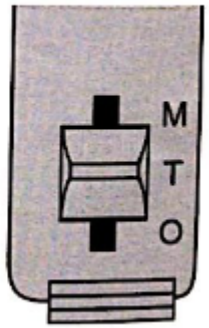
Fine-scale Noise Canceler (FNC)

Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

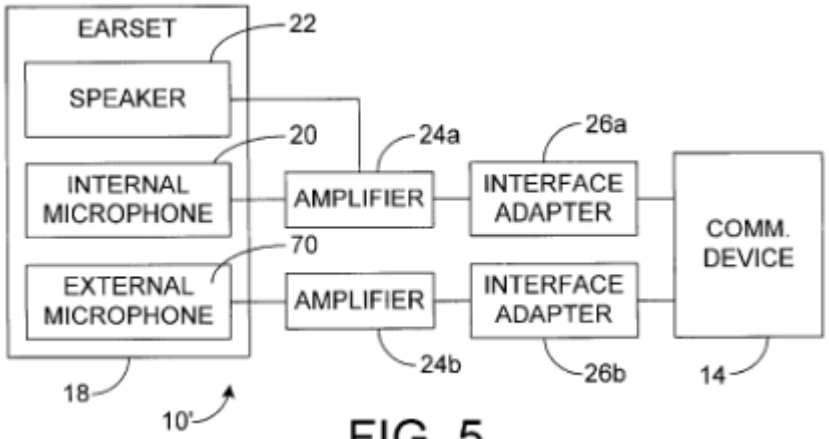
Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

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Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Oticon – DigiFocus Compact User Instructional Brochure at Page 4 **Function switch** **M:** Hearing instrument switched **on** and microphone connected. **Normal use.** **T:** Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise. **O:** Hearing instrument switched **off.**  Anderson '783 at 23:65–24:14 Background noise and competing talker cancellation are achieved using any of a number of techniques and circuits well known in the art, e.g., variations of the Least Mean Squares (LMS) technique as

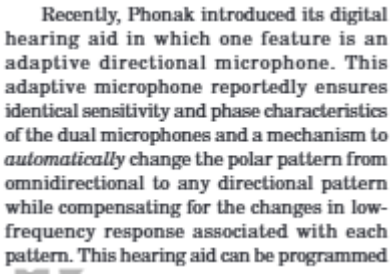
Claims	Prior Art
	implemented in the Motorola DSP56200 LMS filter integrated circuit, or the equivalent function implemented in the general-purpose RPU DSP 948. Note that for all binaural signal processing functions, data contained in replies from the left earpiece (i.e., audio data from the left earpiece microphone) is processed in the RPU DSP 948 by combining said data with other data contained in replies from the right earpiece (i.e., audio data from the right earpiece microphone). The processed audio data transmitted from the RPU to the left and right earpiece speakers via RPU interrogations is therefore a function of data received from both the left and right earpiece microphones via earpiece replies. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes.

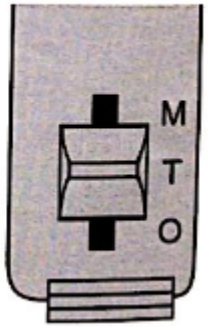
Claims	Prior Art
	Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone. Nemirovski ’379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from

Claims	Prior Art
	the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. Nemirovski ’379 at 9:33–39 The external microphone 70 is disposed on a exterior portion 72 of the housing 30 and is pointed away from the wearer of the earset 18 assembly. The external microphone 70 can be secured to the housing 30 with adhesive and can be disposed in a recess defined by the exterior portion 72. Similar to the internal microphone 20, the external microphone 70 is small (e.g., less than two mm in size). Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. Nemirovski ’379 at 10:8–13 The unique signals generated by the internal and external microphone 20, 70 can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 10′, the system 10′ can be adapted to cancel externally generated noise from the detections of the internal microphone 20. Nemirovski ’379 at 10:14–26 When the user speaks both the internal microphone 20 and the external microphone 70 detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle,

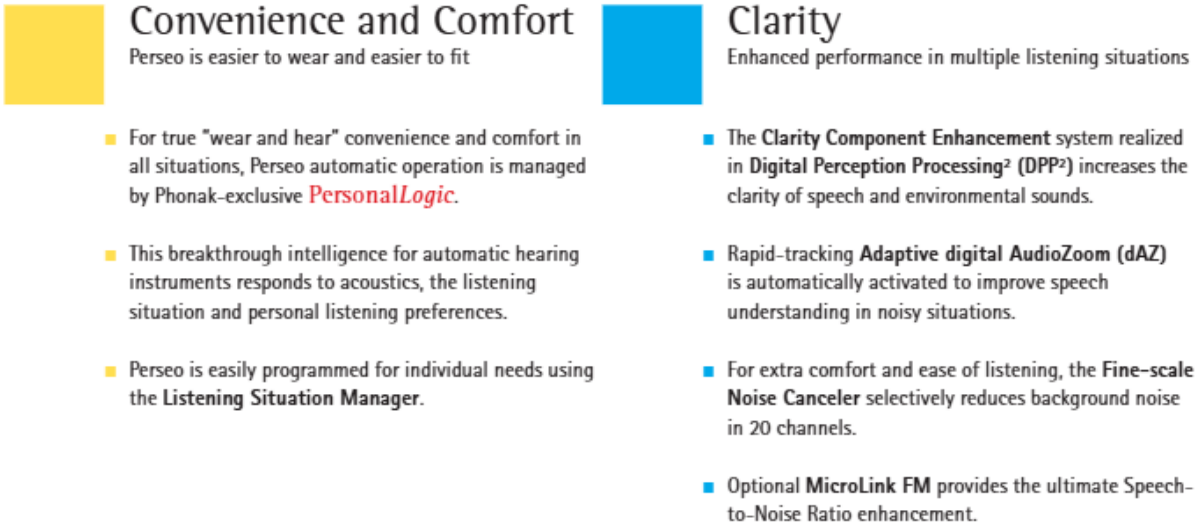
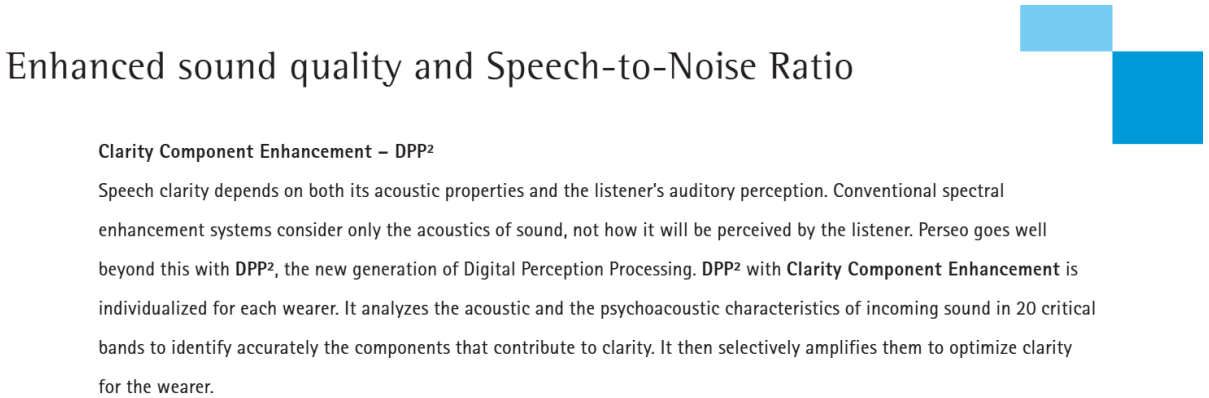
Claims	Prior Art
	machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 70 and the internal microphone 20 detect the noise. Accordingly, the communication device 14 (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 12 by the communication device 14. Nemirovski '379 at 10:27–47 In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or "*", represents a convolution operator): [Equation] $F_{ME}*S+F_{ME}*N=Y_E$ Eq. 2 By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N. *See also* above at claim [11e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[35a] A head-worn personal electronic apparatus as recited in claim 34, wherein the audio input from at least the user picked-up via	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 34, wherein the audio input from at least the user picked-up via the first microphone includes at least a voice input."

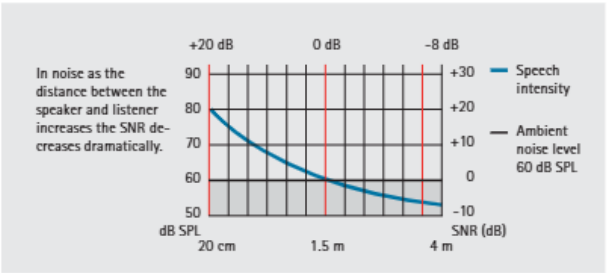
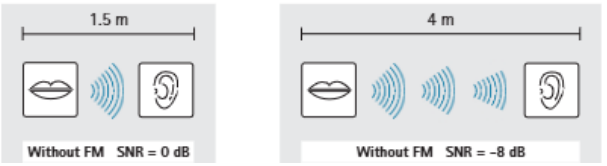
Claims	Prior Art
the first microphone includes at least a voice input, and	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399  Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Phonak – Collections 2002 Product List at Page 7  **Adaptive digital AudioZoom (dAZ)** **Fine-scale Noise Canceler (FNC)** **Active and adaptive separation of speech from noise** The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically. **Fully automatic, fully adaptive** With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR. **Twenty-band noise control** Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal. **Precise and selective** All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected. **Intelligent** The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved. 7

Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4 Function switch **M:** Hearing instrument switched **on** and microphone connected. **Normal use.** **T:** Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise. **O:** Hearing instrument switched **off.**  Anderson ’783 at 26:54–65 Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Jannard ’526 at 18:23–26 Further, the audio device 10A″ can include noise cancellation electronics (not shown) configured to filter wind-generated noise from an audio signal transmitted from the microphone 75. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously

Claims	Prior Art
	enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski '379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. *See also* above at claim [7a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[35b] wherein the environmental audio input associated with the environment includes at least ambient noise.	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "wherein the environmental audio input associated with the environment includes at least ambient noise." Phonak – Perseo Brochure at Page 7  Convenience and Comfort Perseo is easier to wear and easier to fit Clarity Enhanced performance in multiple listening situations For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic. This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences. Perseo is easily programmed for individual needs using the **Listening Situation Manager**. The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds. Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations. For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels. Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement. Phonak – Perseo Brochure at Page 13  Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer.

Claims	Prior Art
	Phonak – Perseo Brochure at p. 14 **Rapid-tracking dAZ** Phonak pioneered the development of directional microphone systems. Not surprisingly, Perseo **dAZ** with new soft switching brings even better performance. Perseo **dAZ** is automatically activated in appropriate situations and immediately reduces noise from beside and behind the wearer. The wearer experiences an immediate reduction of background noise and enhanced speech clarity. Rapid tracking ensures that even fast moving noise sources are controlled. Phonak – Perseo Brochure at Page 15 The perfectly integrated communication system **FM MicroLink** FM technology has allowed thousands of children to participate in standard education settings and to be actively involved in social activities. Phonak pioneered this development and the integration of FM into hearing instruments to produce communication systems that are effective in even the most demanding situations. Since the innovation of miniaturized Phonak **MicroLink** systems, FM has become an important solution also for adults with a high communication need in work and leisure environments. All Perseo BTEs have been designed to incorporate MicroLink with its fully integrated appearance and operation. This integrated system is unique to Phonak. **Speech-to-Noise Ratio (SNR) as a function of distance between speaker and listener**  **Speech-to-Noise Ratio without FM is distance dependent** 

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23



Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Fine-scale Noise Canceler (FNC)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

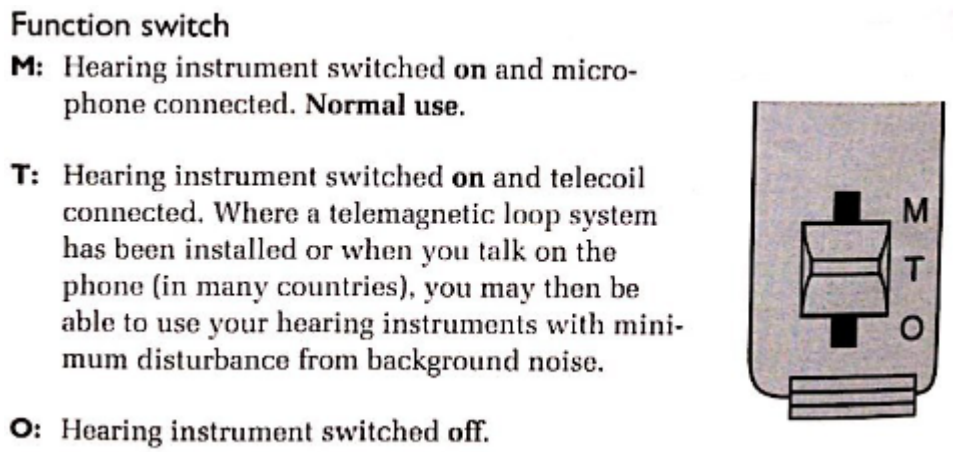
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

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Oticon – DigiFocus Compact User Instructional Brochure at Page 4



Function switch

M: Hearing instrument switched **on** and microphone connected. **Normal use.**

T: Hearing instrument switched **on** and telecoil connected. Where a telemagnetic loop system has been installed or when you talk on the phone (in many countries), you may then be able to use your hearing instruments with minimum disturbance from background noise.

O: Hearing instrument switched **off.**

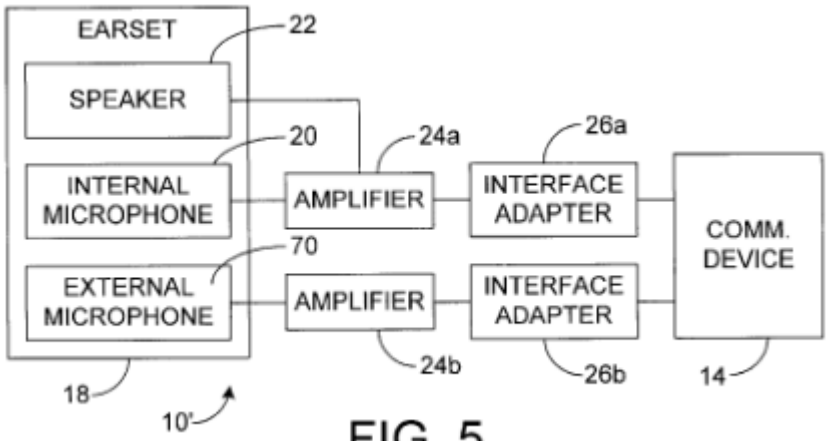
Anderson '783 at 26:54–65

Secondary link audio considerations: In some applications, a user may not wish to transmit signals from the ambient audio environment over the secondary wireless link. In such cases, an auxiliary earpiece

Claims	Prior Art
	microphone which favors the "bone conduction" path over the normal "air path" between the user's mouth and ear can be used. To save space, the auxiliary microphone may be housed in the same miniature case as the primary earpiece microphone. Alternatively, when two earpieces are used, unwanted ambient sources can be "nulled out" in the RPU DSP 948 by digital signal processing techniques well known in the art (e.g., the equivalent of speakerphone circuitry). Jannard ’526 at 18:23–26 Further, the audio device 10A″ can include noise cancellation electronics (not shown) configured to filter wind-generated noise from an audio signal transmitted from the microphone 75. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes.

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson or Jannard do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[36a] A head-worn personal electronic apparatus as recited in claim 34, wherein the first microphone is at least partially in the head-worn personal electronic apparatus and is positioned so as to be directed towards the user of the head-worn personal electronic apparatus, and	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 34, wherein the first microphone is at least partially in the head-worn personal electronic apparatus and is positioned so as to be directed towards the user of the head-worn personal electronic apparatus." Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode). Jannard '526 at 2:26–34 In accordance with another embodiment of at least one of the inventions disclosed herein, an eyeglass frame comprises a support for supporting at least one lens in the path of a wearer's field of view, a first ear stem attached to the support, a second ear stem attached to the support, and at least one microphone supported by at least one of the support, first ear stem, and second ear stem. The microphone is advantageously arranged to face towards the head of a wearer of the eyeglass frame. Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction

Claims	Prior Art
	with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski '379 at FIG. 5  FIG. 5 Nemirovski '379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski '379 at 3:6–23 The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out

Claims	Prior Art
	of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone. Nemirovski '379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Jannard or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[36b] wherein the second microphone is at least partially in the head-worn personal electronic apparatus and is positioned so as to be directed away from the user of the head-worn personal electronic apparatus.	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "wherein the second microphone is at least partially in the head-worn personal electronic apparatus and is positioned so as to be directed away from the user of the head-worn personal electronic apparatus."

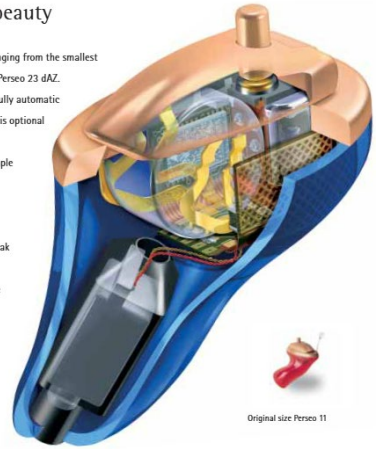
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.



Original size Perseo 11

IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 409

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

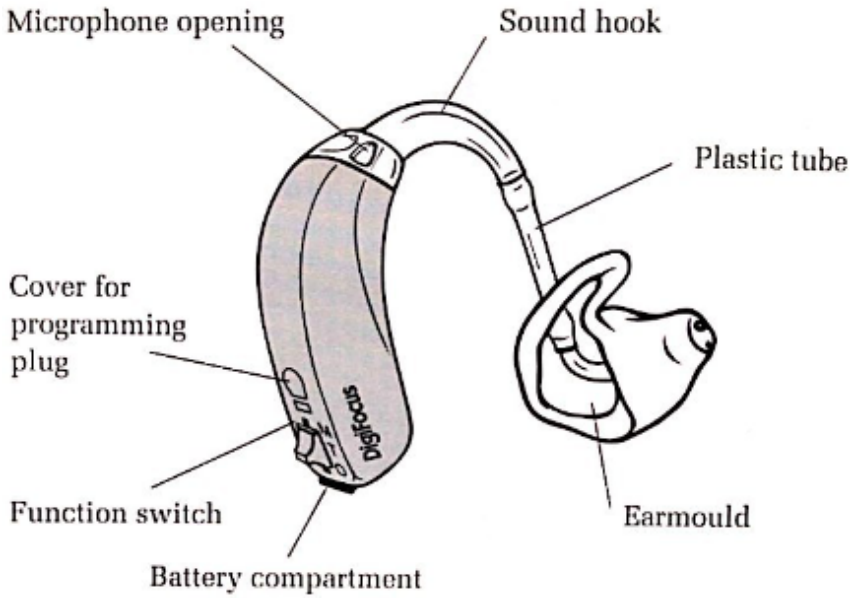
Fine-scale Noise Canceler (FNC)

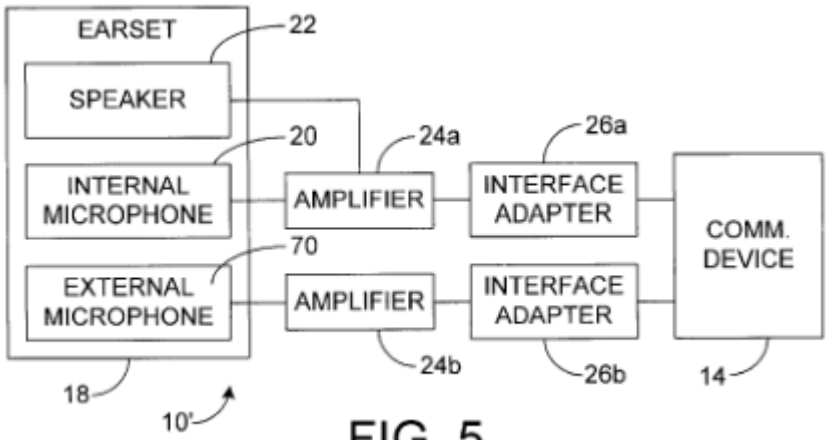
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

7

Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be

Claims	Prior Art
	supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech

Claims	Prior Art
	travels from the mouth of the user and through the air to the external microphone 70. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Jannard or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[37a] A head-worn personal electronic apparatus as recited in claim 36, wherein the audio input from at least the user includes at least a voice input, and	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 36, wherein the audio input from at least the user includes at least a voice input." *See* above at claim [35a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[37b] wherein the environmental audio input includes at least ambient noise.	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "wherein the environmental audio input includes at least ambient noise." *See* above at claim [35b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[38a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn wearable electronic device comprises: a first microphone configured to pick up at least audio input from at least the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn wearable electronic device comprises: a first microphone configured to pick up at least audio input from at least the user." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[38b] a second microphone configured to pickup at least environmental audio input associated with environmental sound in the vicinity of the head-worn personal electronic apparatus, and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone configured to pickup at least environmental audio input associated with environmental sound in the vicinity of the head-worn personal electronic apparatus." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[38c] wherein the electronic circuitry is configured to: determine an environmental sound level based on at least the environmental audio input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to: determine an environmental sound level based on at least the environmental audio input."

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

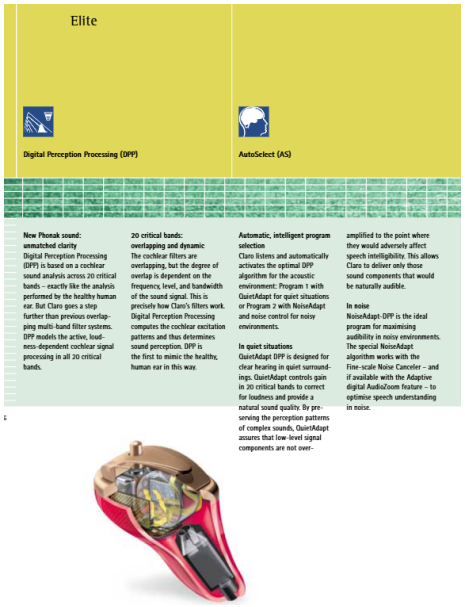
Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

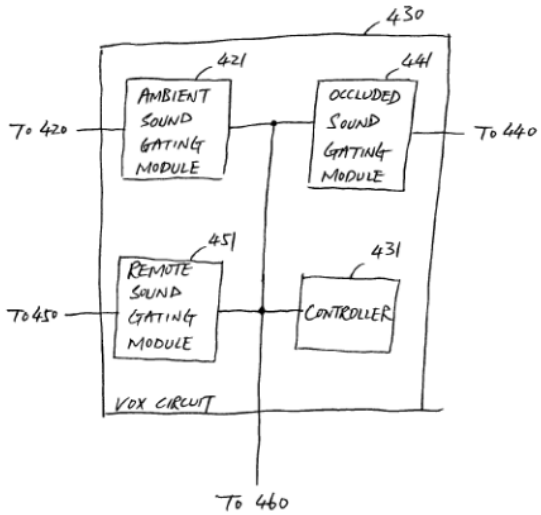
Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 6



Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7  Adaptive digital AudioZoom (dAZ) Fine-scale Noise Canceler (FNC) **Active and adaptive separation of speech from noise** The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically. **Fully automatic, fully adaptive** With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR. **Twenty-band noise control** Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal. **Precise and selective** All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected. **Intelligent** The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved. 7 Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be

Claims	Prior Art
	supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Victorian '313 at FIG. 4B  FIG. 4B Victorian '313 at [0021] FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single ear-level device of FIG. 1. Victorian '313 at [0044] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an ear canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not "isolated" when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440

Claims	Prior Art
	detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110. Victorian ’313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this

Claims	Prior Art
	embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian '313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[38d] amplify at least a portion of the audio input from the first microphone, with the amplification being at least in part dependent on the environmental sound level.	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "amplify at least a portion of the audio input from the first microphone, with the amplification being at least in part dependent on the environmental sound level."

Claims	Prior Art
	Phonak – Perseo Brochure at Page 7

Convenience and Comfort

Perseo is easier to wear and easier to fit

- For true "wear and hear" convenience and comfort in all situations, Perseo automatic operation is managed by Phonak-exclusive PersonalLogic.
- This breakthrough intelligence for automatic hearing instruments responds to acoustics, the listening situation and personal listening preferences.
- Perseo is easily programmed for individual needs using the **Listening Situation Manager**.

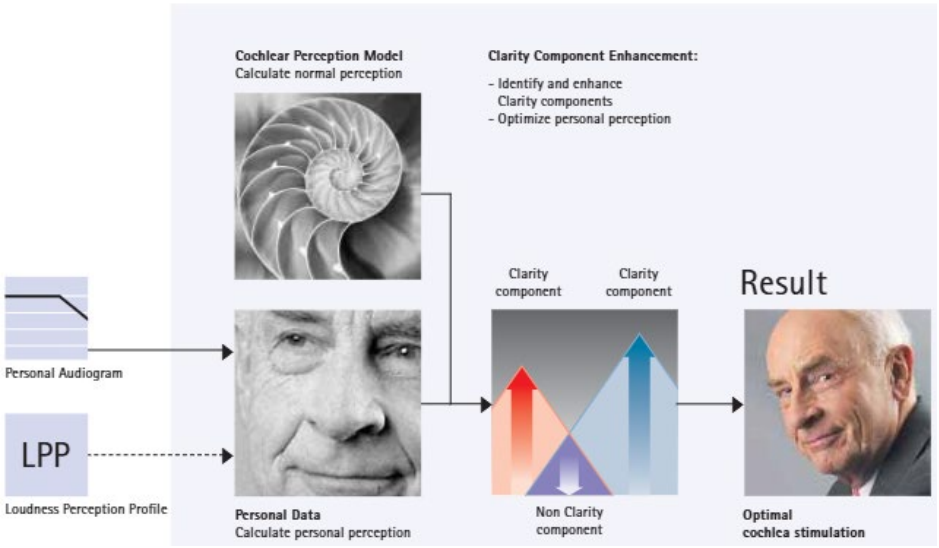
Clarity

Enhanced performance in multiple listening situations

- The **Clarity Component Enhancement** system realized in **Digital Perception Processing² (DPP²)** increases the clarity of speech and environmental sounds.
- Rapid-tracking **Adaptive digital AudioZoom (dAZ)** is automatically activated to improve speech understanding in noisy situations.
- For extra comfort and ease of listening, the **Fine-scale Noise Canceler** selectively reduces background noise in 20 channels.
- Optional **MicroLink FM** provides the ultimate Speech-to-Noise Ratio enhancement.

Phonak – Perseo Brochure at Page 13

Digital Perception Processing² (DPP²)

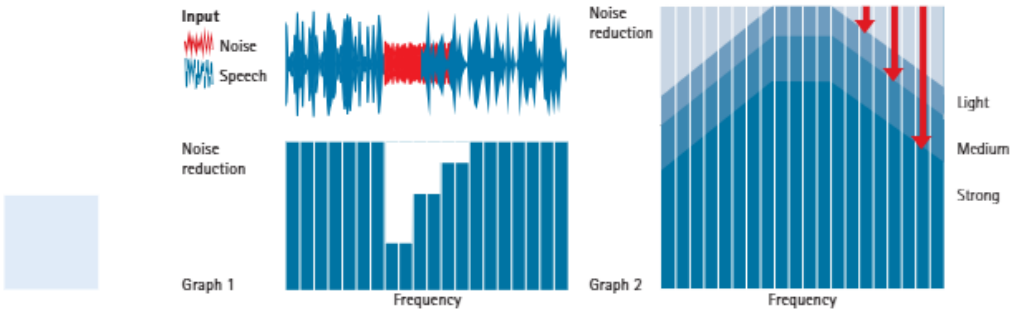


Claims	Prior Art
	Phonak – Perseo Brochure at Page 14

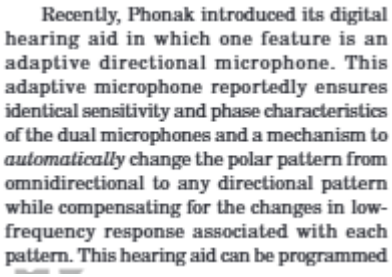
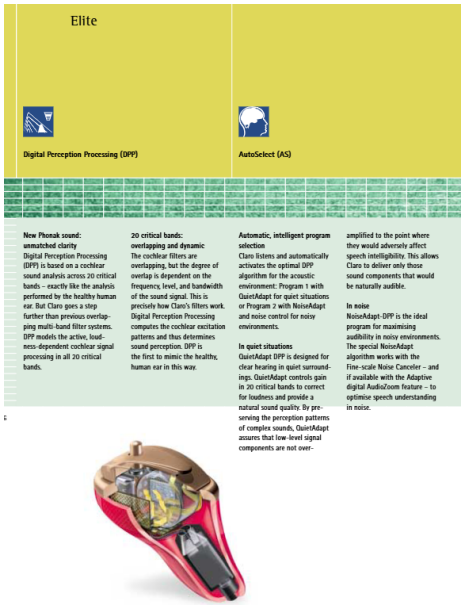
Enhanced comfort in all situations

Fine-scale Noise Canceler in 20 channels

Finely scaled frequency resolution in 20 channels gives the **Fine-scale Noise Canceler (FNC)** the most precise background noise reduction capability. Noise reduction depends on the relative amounts of speech and noise in each channel (graph 1) and the importance of each channel for speech understanding. The **FNC** in Perseo ensures maximum comfort and ease of listening. Annoying background sounds are reduced for comfort while the important sounds of speech are preserved for stress-free listening. The maximum amount of noise reduction can be fitter-programmed to match the individual wearer's tastes and requirements. Thanks to the **dual-base automatic**, separate settings can be made for relatively quiet and noisy situations, ensuring optimal comfort in multiple situations (graph 2).



Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399  Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Phonak – Collections 2002 Product List at Page 6  Elite Digital Perception Processing (DPP) AutoSelect (AS) New Phonak sound: unmatched clarity Digital Perception Processing (DPP) is based on a cochlear sound analysis across 20 critical bands – exactly like the analysis performed by the healthy human ear. But Claro goes a step further than previous overlapping multi-band filter systems. DPP models the active, loudness-dependent cochlear signal processing in all 20 critical bands. 20 critical bands: overlapping and dynamic The cochlear filters are overlapping, but the degree of overlap is dependent on the frequency, level, and bandwidth of the sound signal. This is precisely how Claro's filters work. Digital Perception Processing computes the cochlear excitation patterns and thus determines sound perception. DPP is the first to mimic the healthy, human ear in this way. Automatic, intelligent program selection Claro listens and automatically activates the optimal DPP algorithm for the acoustic environment: Program 1 with QuietAdapt for quiet situations or Program 2 with NoiseAdapt and noise control for noisy environments. In quiet situations QuietAdapt DPP is designed for clear hearing in quiet surroundings. QuietAdapt controls gain in 20 critical bands to correct for loudness and provide a natural sound quality. By preserving the perception patterns of complex sounds, QuietAdapt assures that low-level signal components are not over- amplified to the point where they would adversely affect speech intelligibility. This allows Claro to deliver only those sound components that would be naturally audible. In noise NoiseAdapt-DPP is the ideal program for maximising audibility in noisy environments. The special NoiseAdapt algorithm works with the Fine-scale Noise Canceler – and if available with the Adaptive digital AudioZoom feature – to optimise speech understanding in noise. Oticon – LA Time Article re THE GOODS: The Sound Wave of the Future (Nov. 10 1995) at 1 "Today's amplifiers are not smart," said Peter Hahn, managing director of Oticon's U.S. headquarters in Somerset, N.J. "The digital processor can organize frequencies and manipulate speech signals to separate them from background noise."

Claims	Prior Art
	Oticon – DigiFocus Description at Page 1 Inside each DigiFocus, the digital chips have the same processing power as a fast laptop computer. This power is used to analyse the incoming sound signal, and speech is extracted through a unique system called ASA - Adaptive Speech Alignment. ASA always makes speech audible, amplifying it as much as necessary without making it uncomfortably loud or distorted. Sounds are reproduced very naturally - making speech far easier to understand than with any other hearing instrument on the market. Nemirovski '379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. Nemirovski '379 at 10:8–13 The unique signals generated by the internal and external microphone 20, 70 can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 10′, the system 10′ can be adapted to cancel externally generated noise from the detections of the internal microphone 20. Nemirovski '379 at 10:14–26 When the user speaks both the internal microphone 20 and the external microphone 70 detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle, machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 70 and the internal microphone 20 detect the noise. Accordingly, the communication device 14 (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 12 by the communication device 14.

Claims	Prior Art
	Nemirovski ’379 at 10:27–47 In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal (“S”) and a noise component signal (“N”). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or “*”, represents a convolution operator): [Equation] $F_{ME}*S+F_{ME}*N=Y_E$ Eq. 2 By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N. Victorian ’313 at [0044] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an ear canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not “isolated” when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver

Claims	Prior Art
	receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110. Victorian ’313 at [0045] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440. Victorian ’313 at [0047] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for

Claims	Prior Art
	the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[39a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn wearable electronic device comprises at least a first microphone and a second microphone,	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn wearable electronic device comprises at least a first microphone and a second microphone." *See* above at claim [34a]. A POSITA would have been motivated to combine the teachings of the DHA system and Andrson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[39b] wherein the first microphone is configured to pick up at least audio sound from at least the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "wherein the first microphone is configured to pick up at least audio sound from at least the user." *See* above at claim [35a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[39c] wherein the second microphone is configured to pick up at least ambient noise associated with an environment in the vicinity of the head-worn wearable electronic device, and	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "wherein the second microphone is configured to pick up at least ambient noise associated with an environment in the vicinity of the head-worn wearable electronic device." *See* above at claim [35b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[39d] wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient noise; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient noise." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[39e] modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level." *See* above at claim [38d].

Claims	Prior Art
dependent on the ambient noise level.	A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski or Victorian to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[40a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to receive audio picked-up from the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a first microphone configured to receive audio picked-up from the user." *See* above at claim [35a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[40b] a second microphone configured to receive audio picked-up outward away from the user,	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "a second microphone configured to receive audio picked-up outward away from the user."

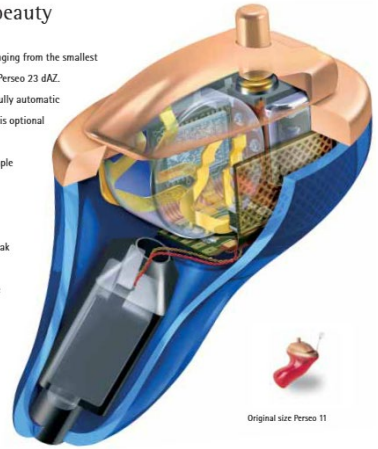
Claims	Prior Art
	Phonak – Perseo Brochure at Page 22

The beauty of high technology

Perseo® ITEs – hidden beauty

The Perseo family includes a full line of ITEs ranging from the smallest Perseo 11 CIC to the elegant dual-microphone Perseo 23 dAZ. All CIC and ITE instruments, like the BTEs, are fully automatic and have Personal*Logic*. A TacTronic switch is optional on all ITE models and easily added after fitting by exchanging battery covers. This is a very simple and practical means of providing extra control and additional programs.

In many countries, ITEs are available with Phonak eShell. These electronically created shells are hypoallergenic, exceptionally durable and more easily retained in the wearer's ear.



Original size Perseo 11

IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 434

Claims	Prior Art
	Phonak – Perseo Brochure at Page 23  Phonak – The Hearing Review Article re Perseo (Sep. 5, 2003) at Page 2 Perseo is a 20-channel digital hearing instrument. The default base-program is optimized for speech understanding in quiet environments. A second base-program with different frequency response settings and optional adaptive directional microphones provides additional support in noisy situations. The acoustic environment is continuously monitored and the appropriate base-program is automatically selected. This dual-base automatic function allows for independent fine tuning in one listening situation without compromising performance in the other. The automatic performance is designed to be easily personalized for the individual needs and preferences of the user.

Claims	Prior Art
	Valente - Effect of type of noise and loudspeaker array on the performance at 399 Recently, Phonak introduced its digital hearing aid in which one feature is an adaptive directional microphone. This adaptive microphone reportedly ensures identical sensitivity and phase characteristics of the dual microphones and a mechanism to *automatically* change the polar pattern from omnidirectional to any directional pattern while compensating for the changes in low-frequency response associated with each pattern. This hearing aid can be programmed Lewis – Speech perception in noise: directional microphones (2004) at Page 429 (2) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the omnidirectional microphone mode; (3) binaural digital Phonak Claro 311 dAZ BTE hearing aids alone in the directional (adaptive) microphone mode; (4) monaural digital Phonak Claro 311 dAZ BTE HA utilized with one Phonak Microlink FM receiver with the EM attenuated (FM-only mode) and one Phonak Claro 311 dAZ BTE HA in the omnidirectional microphone mode worn on the opposite ear; and (5) binaural digital Phonak Claro 311 dAZ BTE hearing aids utilized with binaural Phonak Microlink FM receivers with the EM attenuated (FM-only mode).

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 7



Adaptive digital AudioZoom (dAZ)

Active and adaptive separation of speech from noise
The Speech-to-Noise Ratio (SNR) can only be optimised if the noise is spatially separated from the speech signal with directional microphone technology. As a pioneer in multi-microphone technologies, Phonak introduced AudioZoom, a feature which significantly improves listening and understanding in noisy situations, enhancing user satisfaction. Now, Phonak goes a step further in the refinement of multi-microphone technology with Adaptive digital AudioZoom: it seeks, finds, and suppresses noise automatically.

Fully automatic, fully adaptive
With its unique spatial processing system, Phonak's Adaptive digital AudioZoom can detect noise not only from directly behind, but anywhere in the rear plane. Whether the hearing-impaired person is moving or the location of the noise source changes, the directional characteristic is constantly adapted to the listening situation. In this manner, the Adaptive digital AudioZoom automatically establishes the best possible SNR.

Fine-scale Noise Canceler (FNC)

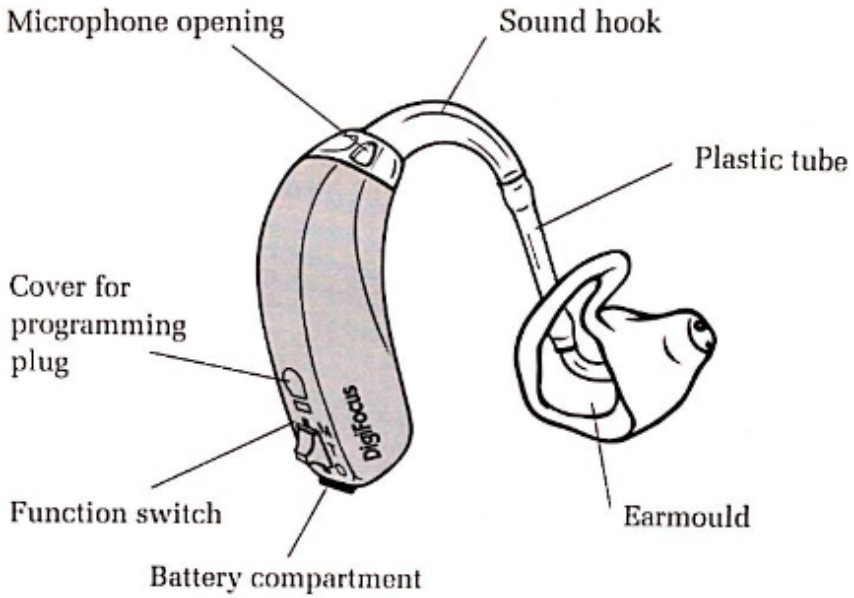
Twenty-band noise control
Claro constantly monitors the Speech-to-Noise Ratio (SNR) in all 20 bands. If a poor SNR is measured in certain bands, the Fine-scale Noise Canceler will reduce gain only in those bands. The result of this precision noise control is a selective and intelligent improvement of the SNR with only a marginal influence on the speech signal.

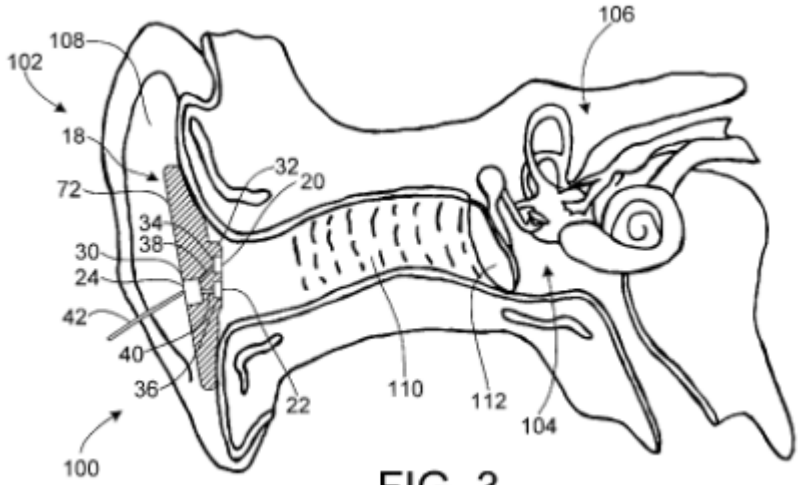
Precise and selective
All frequency bands are separately analysed and controlled. Gain is reduced selectively only in those bands where noise is present. This high-resolution analysis and gain control makes it possible to suppress noise with extremely high accuracy. Neighboring frequency bands without noise remain unaffected.

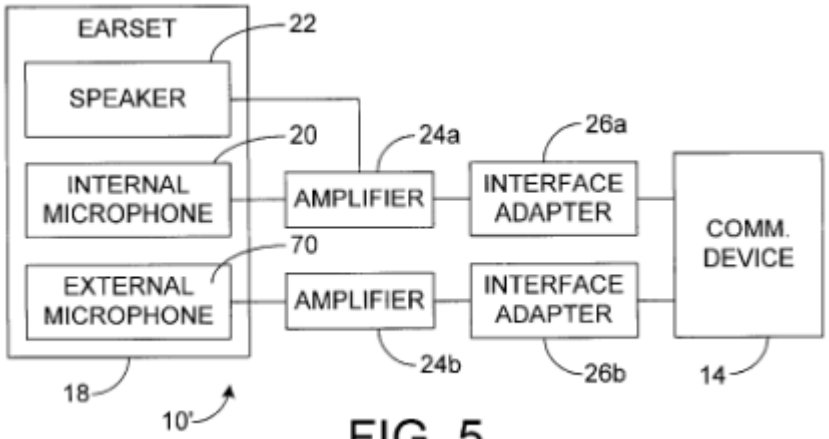
Intelligent
The magnitude of gain reduction is not constant. In frequency bands with higher SNRs and in bands which are important for speech intelligibility, gain is reduced only slightly. This intelligent control algorithm reduces the masking and distractive effects of background noise. Important speech information is preserved.

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Claims	Prior Art
	Oticon – DigiFocus Compact User Instructional Brochure at Page 4  Jannard '526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be

Claims	Prior Art
	supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Nemirovski '379 at FIG. 3  FIG. 3

Claims	Prior Art
	Nemirovski ’379 at FIG. 5  FIG. 5 Nemirovski ’379 at 2:54–56 FIG. 3 is a cross-sectional view of an ear microphone assembly, or earset, disposed with respect to the ear of a person, and the ear also shown in cross-section; Nemirovski ’379 at 2:59–60 FIG. 5 is a block diagram of an ear microphone assembly according to another aspect of the invention; and Nemirovski ’379 at 6:36–41 The microphone 20 is disposed on an interior portion 32 of the housing 30. Accordingly, the interior portion 32 is sized and shaped to position the microphone 20 with respect to the ear 100 and/or ear canal 110 as desired. In one embodiment, the interior portion 32 positions the microphone 20 and/or the speaker 22 in the concha 122. Nemirovski ’379 at 9:19–25 With additional reference to FIG. 5, in another embodiment of the present invention, the earset 18 can include an external microphone 70 in addition to the microphone 20 directed toward the ear canal of the user (also referred to herein as an internal microphone 20). The external microphone 70 detects air

Claims	Prior Art
	pressure changes, or sounds, originating outside the user's ear. Nemirovski ’379 at 9:33–39 The external microphone 70 is disposed on a exterior portion 72 of the housing 30 and is pointed away from the wearer of the earset 18 assembly. The external microphone 70 can be secured to the housing 30 with adhesive and can be disposed in a recess defined by the exterior portion 72. Similar to the internal microphone 20, the external microphone 70 is small (e.g., less than two mm in size). Nemirovski ’379 at 9:45–60 FIG. 5 is a block diagram that illustrates a system 10′ for detecting speech of the user. The system 10′ includes the earset 18 having the internal microphone 20, the speaker 22 and amplifier 24a as described in greater detail above. In addition, the earset 18 of the system 10′ has the external microphone 70. As one skilled in the art will appreciate, the use of the external microphone 70 is optional. The external microphone 70 is disposed on the housing 30 (FIG. 3) and generally points away from the person wearing the earset 18. As a result, the external microphone 70 detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 18, when the speech travels from the mouth of the user and through the air to the external microphone 70. To the extent that DHA systems, alone or in view of Jannard or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[40c] wherein the audio picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the audio picked-up from the second microphone is subtracted from the audio picked-up from the first microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose “wherein the audio picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the audio picked-up from the second microphone is subtracted from the audio picked-up from the first microphone.” *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[41a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a radiation emitter at least partially contained within the head-worn personal electronic apparatus and configured to at least direct radiation towards the user when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn; and	DHA systems, alone, or in view of Jannard or Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a radiation emitter at least partially contained within the head-worn personal electronic apparatus and configured to at least direct radiation towards the user when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn." Jannard '526 at 26:21–27 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the transceiver 114 is adapted to employ a wide variety of technologies, including wireless communication such as RF, IR, ultrasonic, laser or optical, as well as wired and other communications technologies. In one embodiment, a body-LAN radio is employed. Other embodiments can employ a flexible-circuit design. Jannard '526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance. Abreu '960 at Abstract The support structure includes patches, clips, eyeglasses, head mounted gear and the like, containing passive or active sensors positioned at the end of the tunnel with sensing systems positioned on and accessing a physiologic tunnel. Abreu '960 at 2:62–3:12 Included in the present invention are support structures for positioning a sensor on the skin at the end of the tunnel. The present invention discloses devices directed at measuring brain temperature, brain function, metabolic function, hydrodynamic function, hydration status, hemodynamic function, body chemistry and the like. The components include devices and methods for evaluating biological parameters using patches, clips, eyeglasses, head mounted gear and the like with sensing systems adapted to access physiologic tunnels to provide precise and clinically useful information about the physiologic status of the

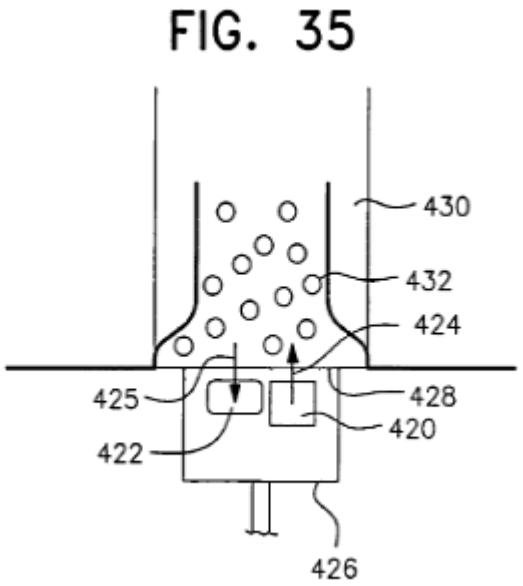
Claims	Prior Art
	wearer and for enhancing the safety and performance of said wearer, and helping to enhance and preserve the life of said wearer by providing adequate reporting means and alert means relating to the biological parameter being monitored. Other components provide for producing direct or indirect actions, acting on another device, or adjusting another device or article of manufacture based on the biological parameter measured. Abreu ’960 at 44:48–51 The support structure includes at least one radiation source from infrared to visible light which interacts with the substance being measured at the BTE tunnel and a detector for collecting the resulting radiation. Abreu ’960 at 44:61–67 The present invention includes means for directing preferably near-infrared energy into the surface of the skin at the end of the BTE tunnel, means for analyzing and converting the reflectance or back scattered spectrun into the concentration of the substance measured and support means for positioning the light source and detector means adjacent to the surface of the skin at the BTE tunnel site. Luo ’446 at [0036] The invention generally relies on shining light having a particular wavelength on human skin as well as the reflectance properties of human skin. The surface reflection of the human skin occurs at the surface of the epidermis and is approximately independent of the lighting wavelength and independent of human race. In the infrared (IR) range the wavelengths that can be used are greater than 800 nano-meters and the surface reflectance of human skin is more or less constant and close to 50%. The same hold true for orange and red light in the visible light range, namely wavelengths between approximately 600 and 800 nano-meters can be used, although the surface reflectance for darker skin is reduced for shorter wavelengths. Surface reflectance for dark skin is approximately 25% at 700 nano-meters. In the embodiments shown herein, the energy of choice in the sensor unit 18 is preferably infrared (IR) energy. However, long wavelength visible light energy can also be used as discussed below. Luo ’446 at [0038] The optical window 56 is placed at a certain location on the shell 58 of the hearing aid 10. The location of the optical window 56 depends on whether the hearing aid is a BTE, ITE, ITC or CIC hearing aid. In addition, the location is chosen to minimize the distance between the optical window 56 and the skin 64. For optimal reflection of IR signals back through the optical window, the hearing aid user's skin (i.e. the

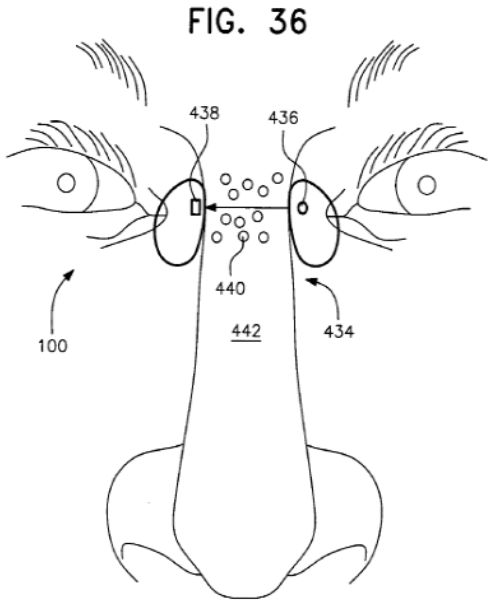
Claims	Prior Art
	reflecting surface) is required to be immediately over the optical window 56; otherwise the reflected IR energy will not be reflected back towards the reception unit 62. The optical window 56 is typically a small window having a diameter of approximately 1 mm for example. The optical window 56 can be made from IR grade glass or other suitable material that allows for the passage of infrared energy (a different material would be used if visible light is used rather than infrared light). The optical window 56 should be kept clean at all times in order to prevent the emitted infrared energy from being reflected back due to dirt and the like that may accumulate on the optical window 56 over time. The location of the optical window 56 will be discussed in further detail below. Irvin '332 at 6:3–33 FIG. 4C illustrates the power control circuit 210 coupled with a contact switch functioning as the sensor 220. The contact switch includes a first and second contact connected to the power control circuit 210, and connecting member 222 actuated when the user places the wireless headset 100 on their body. When actuated, the connecting member 222 places the first and second contacts of sensor 220 in electrical connection. Ideally, the sensor 220, when configured as a contact sensor, is placed in the earpiece 130, in the retaining member 140, or in the retaining pad 150 shown in FIG. 1B. Preferably, when sensor 220 is configured as a contact-based proximity sensor, the contacting element of sensor 220 is sealed in a manner that minimizes contaminant ingress. In some embodiments, the contact sensor may be actuated by the user resiliently expanding the retaining member 140 or other element of the wireless headset 100 for mounting upon their body. FIG. 4D illustrates the power control circuit 210 coupled with a proximity sensor 220 not based on mechanical actuation. The proximity sensor 220 may be an active or passive IR sensor, may be a resistive or other type of thermal sensor, or may be an acoustic sensor. As a proximity sensor, sensor 220 provides indication that a user has placed the wireless headset 100 upon their body. As such, any of the foregoing or other presence sensing techniques may be used to implement sensor 220. Preferably, when configured as a proximity sensor, sensor 220 requires minimal power and provides a sensor output signal based on proximity to the user's body. Depending upon the sensing technology chosen, the control circuit 210 may be required to periodically stimulate sensor 220 to generate a detection response. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Jannard or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to

Claims	Prior Art
	combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[41b] a radiation detector at least partially contained within the head-worn personal electronic apparatus and configured to detect radiation reflected back from the user when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn,	DHA systems, alone or in view of Jannard or Abreu or Luo or Irvin, expressly or inherently disclose "a radiation detector at least partially contained within the head-worn personal electronic apparatus and configured to detect radiation reflected back from the user when at least the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn." *See* above at claim [13a], [13b], and [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Jannard or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[41c] wherein the radiation detector is operatively connected to the electronic circuitry, and wherein electronic circuitry within the head-worn personal electronic apparatus is configured to facilitate determining whether the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn by the user based on at least the detected radiation.	DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, expressly or inherently disclose "wherein the radiation detector is operatively connected to the electronic circuitry, and wherein electronic circuitry within the head-worn personal electronic apparatus is configured to facilitate determining whether the head-worn personal electronic apparatus of the head-worn personal electronic apparatus is being worn by the user based on at least the detected radiation." *See* above at claim [13a], [13b], and [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
42. A head-worn personal electronic apparatus as recited in claim 41, wherein the detected	DHA systems, alone or in view of Jannard or Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 41, wherein the detected radiation is at least infrared radiation."

Claims	Prior Art
radiation is at least infrared radiation.	*See* above at claim [41b]. To the extent that DHA systems, alone or in view of Jannard or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[43a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data,	DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data." *See* above at claim [13a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[43b] wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector.	DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, expressly or inherently disclose "wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector." *See* above at claim [13b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
44. A head-worn personal electronic apparatus as recited in claim 43, wherein the electronic	DHA systems, alone or in view of Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 43, wherein the electronic circuitry is configured to

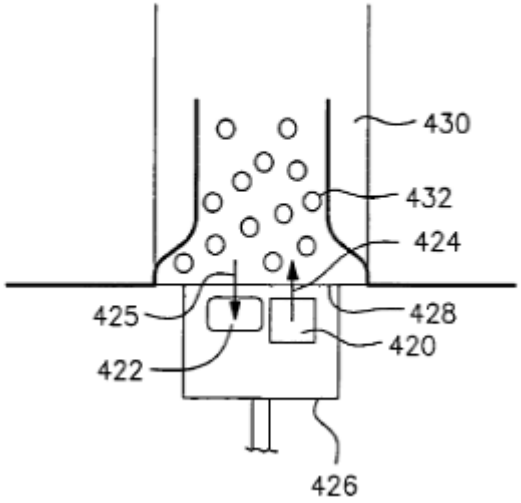
Claims	Prior Art
circuitry is configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data.	determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data." *See* above at claim [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Abreu or Luo or Irvin do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
45. A head-worn personal electronic apparatus as recited in claim 44, wherein the radiation detector comprises a photodetector.	DHA systems, alone or in view of Jannard or Abreu, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 44, wherein the radiation detector comprises a photodetector." Jannard '526 at 26:21–27 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the transceiver 114 is adapted to employ a wide variety of technologies, including wireless communication such as RF, IR, ultrasonic, laser or optical, as well as wired and other communications technologies. In one embodiment, a body-LAN radio is employed. Other embodiments can employ a flexible-circuit design. Jannard '526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance.

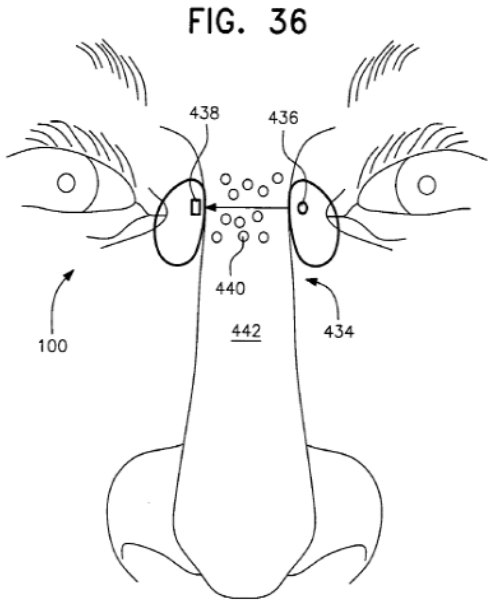
Claims	Prior Art
	Abreu ’960 at FIG. 35 

Claims	Prior Art
	Abreu ’960 at FIG. 36  Abreu ’960 at 45:60–46:13 In accordance with FIG. 35 a schematic view of one preferred reflectance measuring apparatus of the present invention is shown. FIG. 35 shows a light source 420 such as an infrared LED and a photodetector 422 located side-by-side and disposed within support structure 426 such as a medial canthal pad or modified nose pads of eyeglasses directing radiation 424 at the BTE tunnel 430 with said light source 420 laying in apposition to the skin 428 at the BTE tunnel 430. The light source 420 delivers the radiation 424 to the skin 428 at the BTE tunnel which is partially absorbed according to the interaction with the substance 432 being measured resulting in attenuated radiation 425. Part of the radiation 424 is then absorbed by the substance 432 and the resulting radiation 425 emitted from BTE tunnel 430 is collected by the photodetector 422 and converted by a processor into the blood concentration of the substance 432. Thin skin 428 is the only tissue interposed between radiation 424, 425 and the substance 432 being measured. The concentration of the substance 432 is accomplished by detecting the magnitude of light attenuation collected which is caused by the absorption signature of the substance being measured.

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Jannard or Abreu, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
46. A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises a force sensor configured to receive a user input based at least on sensing force applied to the force sensor.	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises a force sensor configured to receive a user input based at least on sensing force applied to the force sensor." *See* above at claim [14a]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, and 14. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
47. A head-worn personal electronic apparatus as recited in claim 46, wherein the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device.	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 46, wherein the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device." *See* above at claim [14b]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, and 14. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

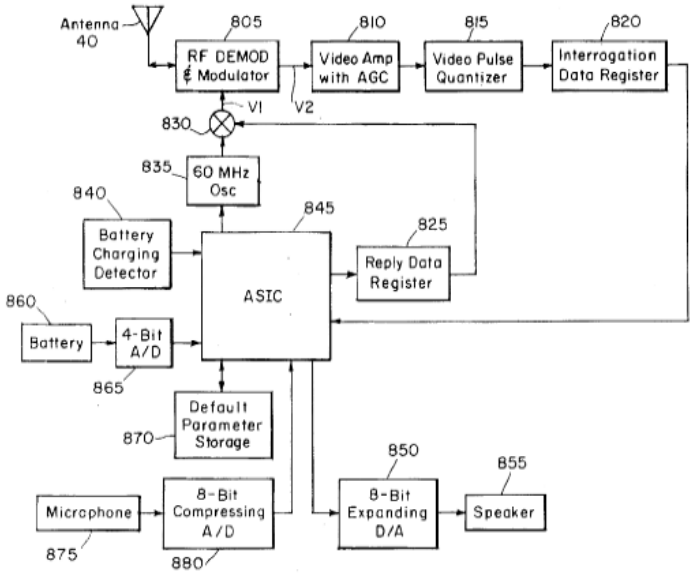
Claims	Prior Art
48. A head-worn personal electronic apparatus as recited in claim 47, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the audio output being provided to the user.	DHA systems, alone or in view of Jannard or Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 47, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the audio output being provided to the user." *See* above at claim 27. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, 14, and 27. To the extent that DHA systems, alone or in view of Jannard or Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
49. A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a light detector configured to capture light data, the light detector being electrically connected to the electronic circuitry, and the electronic circuitry being configured to process at least the light data captured by the light detector.	DHA systems, alone or in view of Jannard or Abreu or Luo, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a light detector configured to capture light data, the light detector being electrically connected to the electronic circuitry, and the electronic circuitry being configured to process at least the light data captured by the light detector." Jannard '526 at 26:21–27 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the transceiver 114 is adapted to employ a wide variety of technologies, including wireless communication such as RF, IR, ultrasonic, laser or optical, as well as wired and other communications technologies. In one embodiment, a body-LAN radio is employed. Other embodiments can employ a flexible-circuit design. Jannard '526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance.

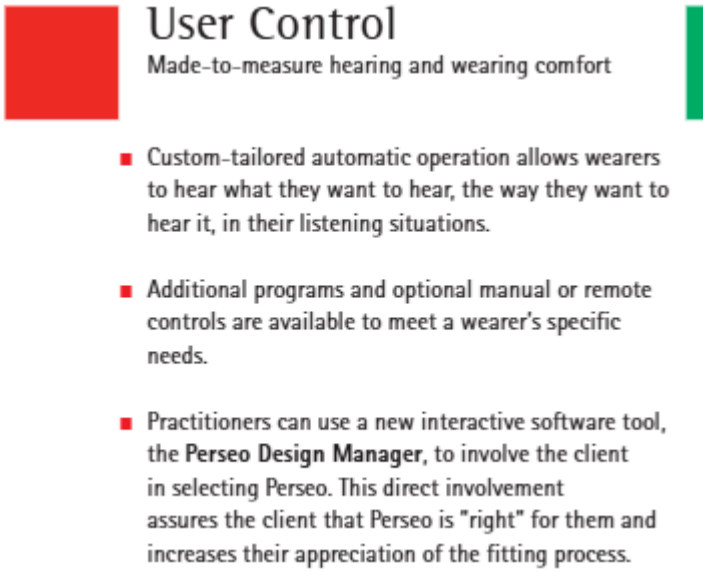
Claims	Prior Art
	Abreu ’960 at FIG. 35 FIG. 35 

Claims	Prior Art
	Abreu ’960 at FIG. 36  Abreu ’960 at 22:42–46 FIG. 35 is a schematic view of another preferred embodiment of the present invention using spectral reflectance. FIG. 36 is a schematic view of a person showing another preferred embodiment in accordance with the present invention using spectral transmission. Abreu ’960 at 45:60–46:13 In accordance with FIG. 35 a schematic view of one preferred reflectance measuring apparatus of the present invention is shown. FIG. 35 shows a light source 420 such as an infrared LED and a photodetector 422 located side-by-side and disposed within support structure 426 such as a medial canthal pad or modified nose pads of eyeglasses directing radiation 424 at the BTE tunnel 430 with said light source 420 laying in apposition to the skin 428 at the BTE tunnel 430. The light source 420 delivers the radiation 424 to the skin 428 at the BTE tunnel which is partially absorbed according to the interaction with the substance 432 being measured resulting in attenuated radiation 425. Part of the radiation 424 is then absorbed by the substance 432 and the resulting radiation 425 emitted from BTE tunnel 430 is collected by the photodetector 422 and converted by a processor into the blood concentration of the substance 432. Thin skin 428 is the only tissue interposed between radiation 424, 425 and the substance 432 being

Claims	Prior Art
	measured. The concentration of the substance 432 is accomplished by detecting the magnitude of light attenuation collected which is caused by the absorption signature of the substance being measured. Abreu '960 at 46:28–41 FIG. 36 shows a schematic view of a person 100 wearing a support structure 434 and light source 436 and detector 438 adapted to measure biological parameters using spectral transmission means. The light source 436 and photodetector 438 are positioned diametrically opposed to each other so that the output of the radiation source 436 goes through the nasal interface 442 containing the substance 440 being measured before being received by the detector 438. Photodetector 438 collects the resulting transmitted radiation which was directed through the nasal interface 442. A variety of LEDs and optical fibers disposed within the support structure 434 such as the medial canthal pads, nose pads and frames of eyeglasses are preferably used as a light delivery for the light source 436 and the light detector 438. Luo '446 at [0036] The invention generally relies on shining light having a particular wavelength on human skin as well as the reflectance properties of human skin. The surface reflection of the human skin occurs at the surface of the epidermis and is approximately independent of the lighting wavelength and independent of human race. In the infrared (IR) range the wavelengths that can be used are greater than 800 nano-meters and the surface reflectance of human skin is more or less constant and close to 50%. The same hold true for orange and red light in the visible light range, namely wavelengths between approximately 600 and 800 nano-meters can be used, although the surface reflectance for darker skin is reduced for shorter wavelengths. Surface reflectance for dark skin is approximately 25% at 700 nano-meters. In the embodiments shown herein, the energy of choice in the sensor unit 18 is preferably infrared (IR) energy. However, long wavelength visible light energy can also be used as discussed below. Luo '446 at [0038] The optical window 56 is placed at a certain location on the shell 58 of the hearing aid 10. The location of the optical window 56 depends on whether the hearing aid is a BTE, ITE, ITC or CIC hearing aid. In addition, the location is chosen to minimize the distance between the optical window 56 and the skin 64. For optimal reflection of IR signals back through the optical window, the hearing aid user's skin (i.e. the reflecting surface) is required to be immediately over the optical window 56; otherwise the reflected IR energy will not be reflected back towards the reception unit 62. The optical window 56 is typically a small window having a diameter of approximately 1 mm for example. The optical window 56 can be made from IR grade glass or other suitable material that allows for the passage of infrared energy (a different material

Claims	Prior Art
	would be used if visible light is used rather than infrared light). The optical window 56 should be kept clean at all times in order to prevent the emitted infrared energy from being reflected back due to dirt and the like that may accumulate on the optical window 56 over time. The location of the optical window 56 will be discussed in further detail below. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Luo to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, and 41. To the extent that DHA systems, alone or in view of Jannard or Abreu or Luo, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
50. A head-worn personal electronic apparatus as recited in claim 49, wherein the light detector comprises a photodetector.	DHA systems, alone or in view of Jannard or Abreu, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 49, wherein the light detector comprises a photodetector." *See* above at claim 45. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, and 45. To the extent that DHA systems, alone or in view of Jannard or Abreu, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
51. A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus is configured to provide audio alerts to the user via the at least one speaker.	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus is configured to provide audio alerts to the user via the at least one speaker." Phonak – Fitting Guide for MLxi and MLxi Baha at Page 5 Beep type MLxi/MLxi Baha plays a confirmation beep at start up or when the system synchronizes with a transmitter. You can change the type of beep tone using the drop-down menu.

Claims	Prior Art
	Anderson ’783 at FIG. 8  FIG. 8 Anderson ’783 at 3:40–41 FIG. 8 is a block diagram showing details of an earpiece for a full-featured preferred embodiment of the invention. Anderson ’783 at 24:15 Unsolicited information: Anderson ’783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948.

Claims	Prior Art
	Jannard '526 at 20:1–5 For example, but without limitation, where the source device B is a cellular phone, the user U can listen to the audio output of the cellular phone, such as the voice of a caller, through sound transducers in the audio device 10C. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
52. A head-worn personal electronic apparatus as recited in claim 51, wherein the audio alerts are dependent on one or more user preference settings.	DHA systems, alone of in view of Anderson or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 51, wherein the audio alerts are dependent on one or more user preference settings." Phonak – Perseo Brochure at Page 7  User Control Made-to-measure hearing and wearing comfort ■ Custom-tailored automatic operation allows wearers to hear what they want to hear, the way they want to hear it, in their listening situations. ■ Additional programs and optional manual or remote controls are available to meet a wearer's specific needs. ■ Practitioners can use a new interactive software tool, the **Perseo Design Manager**, to involve the client in selecting Perseo. This direct involvement assures the client that Perseo is "right" for them and increases their appreciation of the fitting process.

Claims	Prior Art
	Anderson ’783 at 24:15 Unsolicited information: Anderson ’783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948. Jannard ’526 at 16:23–34 In the illustrated embodiment, the buttons 73a and 73b for controlling the volume of the sound output from the speakers 14A′, 16A′. For example, the button 73a can be used for increasing volume and the button 73b can be used for decreasing volume. Alternatively, the button 73b can be for increasing volume and the button 73a can be for decreasing volume. When a wearer of the audio device 10A′ presses one of the buttons 73a, 73b, a simple on-off signal can be transmitted to the device 286. The device 286 can be configured to interpret the on-off signals from the buttons 73a, 73b as volume control signals and adjust the volume to the speakers 14A′, 16A′ accordingly. Jannard ’526 at 24:20–27 In one illustrative and non-limiting embodiment, the button 152 can be used to control volume. For example, by pressing on the forward portion 156, a contact can be made causing the transceiver 114 or the interface 116 to increase the volume of the speakers 14D, 16D. Additionally, by pressing on the rearward portion 158 of the button 152, the transceiver 114 or interface 116 could lower the volume of the speakers 14D, 16D. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with

Claims	Prior Art
	the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
53. A head-worn personal electronic apparatus as recited in claim 17, wherein the at least one speaker is configured to be at least partially inserted into an ear of the user.	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the at least one speaker is configured to be at least partially inserted into an ear of the user." Phonak – Perseo Brochure at p. 8  Valente - Effect of type of noise and loudspeaker array on the performance at 398  Differences in performance between omnidirectional and directional microphones were evaluated between two loudspeaker conditions (single loudspeaker at 180°; diffuse using eight loudspeakers set 45° apart) and two types of noise (steady-state HINT noise; R-Space™ restaurant noise). Twenty-five participants were fit bilaterally with Phonak Perseo hearing aids using the manufacturer's recommended procedure. After wearing the hearing aids for one week, the Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF

Claims	Prior Art
	transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Jannard '526 at 1:34–37 The audio output for many of these systems is typically directed to the wearer through the use of transducers physically positioned in or covering the ear, such as earphones and headphones. Jannard '526 at 10:51–57 Further, some wearers of the audio device 10A′ may prefer to press the speakers 14A′, 16A′ into contact with the outer surfaces of their ears. For example, some users may desire to experience to loudest possible volume from the speakers 14A′, 16A′. Thus, by pressing the speakers 14A′, 16A′ against their ears, the perceived volume of the sound emitted from the speakers 14A′, 16A′ will be the greatest. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[54a] A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a microphone at least partially internal to the head-worn personal electronic apparatus,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: a microphone at least partially internal to the head-worn personal electronic apparatus." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[54b] wherein the head-worn personal electronic device is configured to receive a voice input via the microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation.	DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, expressly or inherently disclose "wherein the head-worn personal electronic device is configured to receive a voice input via the microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation." Anderson '783 at 22:11–61 Stand-alone earpiece operation: If an earpiece cannot communicate with an RPU for any reason, stand-alone earpiece operation as a CIC hearing aid is generally desirable. Examples include occasions where the RPU must be turned off by pushing the RPU power-off key 78 (e.g., situations where operation of wireless transmitters is prohibited such as aircraft takeoffs and classified military briefings), RPU battery or other RPU electronics failure, misplacement of an RPU, and primary wireless link 17 interference. Note that a microphone located in the earpiece 10 is preferred to a body-worn microphone, since only an earpiece microphone 12 can be used to provide the desired default CIC hearing aid function (earpiece microphones also provide superior audio localization capabilities as the microphone moves with the user's head, and provide reduced wind and clothing noise compared with body-worn microphones). Parameters that control the stand-alone earpiece operation are stored in the earpiece CIC default parameter storage 870, which is accessed by the ASIC 845 to implement the desired stand-alone functions as described earlier. CIC default parameters include, but are not limited to, settings for the nominal and maximum allowable earpiece volume. Other information contained in the default parameter storage 870 includes, but is not limited to, the communication address and L/R (left/right) selection bit. To ensure that the CIC default parameter storage information cannot be changed inadvertently, changes are only allowed when the earpiece is not in use as, for example, when the earpiece battery 860 is being charged. A battery charging detector 840, that is sensitive to the high-level magnetic induction fields commonly used for

Claims	Prior Art
	earpiece battery charging, indicates to the ASIC 845 that the default parameter storage 870 values may be changed upon receipt of an appropriate command from the RPU. Specifically, to change information in the earpiece CIC default parameter storage 870, a user first activates the earpiece battery charging system, e.g., by placing the earpiece in a recharger/storage tray well known in the art. The RPU keyboard 946 is then used to enter a command to the DSP 948 that indicates the user's desire to change information in the default parameter storage 870, followed by the new information to be stored. The DSP 948 causes the RPU to transmit an interrogation with a reserved address code to which all earpieces being charged within the communication range of the RPU are receptive. The information that is to ultimately be loaded into the earpiece default parameter storage 870 is then transmitted by the RPU using the auxiliary bit field of many successive interrogations, said interrogations are received by the earpiece, and the desired information is loaded into the CIC default parameter storage 870 under control of the ASIC 845. Anderson ’783 at 25:15–27 To obtain solicited information, user requests are processed in a variety of means well known in the art, including inputs via one or more pushbuttons on the RPU and voice recognition computer programs implemented in the RPU electronics. Voice recognition can be used almost exclusively, if desired, thereby eliminating nearly all RPU pushbutton controls. Voice commands recognized by the RPU can cause the RPU to initiate actions such as changing hearing aid parameter settings (e.g., volume), or activate appliances and environmental controls (e.g., aids for the physically handicapped) in a distributed wireless network via the secondary wireless link. Anderson ’783 at 25:29–30 The verbal commands are received as audio signals by the earpiece 10 and transmitted to the RPU 16. Jannard ’526 at 25:53–26:20 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices include at least two banks of microphones, with one bank acting as a speaker of received and one bank providing an ambient noise-cancellation function. The microphone banks can be positioned at any suitable location or combination of locations (e.g., on the audio device, within the audio device, opposing sides of the audio device, or the like). In one embodiment, automatic switching of the speaking-microphone and noise-canceling-microphone banks' functions advantageously enhances ease of use. In a further embodiment, the microphone banks can be arranged in an array to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise for the purpose of voice recognition. For example, in one embodiment, such microphone banks can include ASIC-based noise-

Claims	Prior Art
	canceling technology, such as is available in chips from Andrea Electronics Corporation (AEC), to enable voice recognition in ambient noise up to about 130 Db or more. In another embodiment, microphone banks can be arranged in any suitable combination of linear or non-linear arrays to be used in conjunction with algorithms to discern, reduce, and/or eliminate noise in any suitable manner. In another embodiment, audio/proximity sensors can advantageously trigger the appropriate functionality in a specific bank. In another embodiment, a noise-canceling microphone can be provided in connection with a cord or other microphones described above. For example, without limitation, a series of miniature microphones can be supported down a cord from the audio device, separated by desired distances, and aimed in different directions. In another implementation, one or more of the microphones can be for verbal input from the user, and one or more others of the microphones, or the same microphone, can also be for noise-cancellation purposes. Jannard '526 at 30:53–31:2 With reference to FIG. 9A, the interface electronics 116 can include audio or "aural" menus that can be selected by user. For example, a user can initiate an audio menu by depressing the button 150 (FIGS. 10–12). Upon initiation of the audio menus, the interface electronics 116 can send an audio signal to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E including a humanoid voice. The voice signal can indicate that a first menu option is available. For example, but without limitation, the first menu choice can be to initiate a call. Thus, when the user pushes the button 150 the first time, the user will hear the words "initiate a call," emanating from the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. If the user wishes to initiate a call, the user can depress the button 150 again which will send the appropriate AT command to the transceiver 114 so as to transmit the proper AT code to the cellular phone source device S, B (FIG. 8). Jannard '526 at 32:4–13 For example, if a user chooses a voice command option, the user could issue commands such as, for example, but without limitation, "phonebook" or "call alpha." With a source device such as a cellular phone, that has a speech recognition engine and that is properly trained to recognize the voice of the user, the user can automatically enter the phonebook mode or directly call the phonebook listing "alpha," of course, as is apparent to one of ordinary skill in the art, such a voice command protocol could be used to issue other commands as well. Jannard '526 at 32:14–19 In another alternative, the interface electronics 116 can include a speech recognition engine and audio

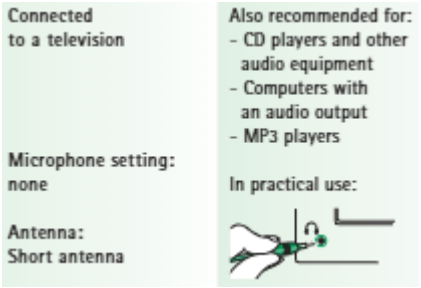
Claims	Prior Art
	menus. In this alternative, the interface electronics 116 can recognize speech from the user, convert the speech to AT commands, and control this source device using a standard AT command protocol. Nemirovski '379 at 4:27–41 The communications system 10 includes an ear microphone assembly, or earset 18, having a microphone 20 and a speaker 22 supported by a housing 30 (FIG. 3). The physical arrangement of the earset 18 will be described in more detail below. The microphone 20 is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 20 converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24. Next, the amplified electrical signal is received by the communication device 14 via an appropriate interface adapter 26. The interface adapter 26 can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 14. Seshadri '839 at [0042] Input module 128, which may be a keypad, touch screen, voice recognition circuit, or other like user interfaces, receives user commands and produces digital command messages 124 there from. Such digital command messages 124 includes, but are not limited to, packet size, synchronization information, frequency hopping initiation information, timeslot allocation information, link establishment information, piconet address information, fast-forward, play, pause, volume adjust, record, stop and rewind. Seshadri '839 at [0047] Detachable ear-piece 156 and microphone 154 may have on-chip operations to support call conferencing, call waiting, flash, and other features associated with telephones. These functions may me accessed and reviewed by a user interface 158 and display 160 within host device 152 or a user interface and display located on either detachable ear-piece 156 or microphone 154. The user interface and display, located on either the host device or detachable ear-piece 156 and microphone 154 may have a display and button(s) that may be used to program device, perform directory functions including selecting number to call, view caller ID, initiate call waiting, or initiate call conferencing. Additionally, circuitry within the ear-piece 156 and microphone 154 may enable voice activated dialing. The actual voice recognition could be performed within ear-piece 156, microphone 154, or host device 152. Thus, ear-piece 156 and microphone 154 may act to initiate calls and receive calls.

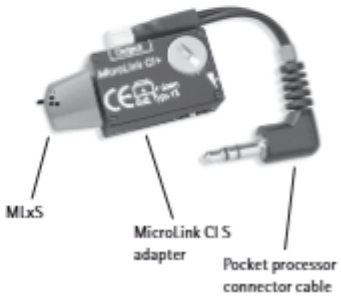
Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard or Nemirovski or Seshadri to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
57. A head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: an array of microphones configured for directional audio pickup.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the head-worn personal electronic apparatus comprises: an array of microphones configured for directional audio pickup." *See* above at claim [34a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
58. A head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output includes music that was wirelessly received via the wireless receiver.	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output includes music that was wirelessly received via the wireless receiver." Phonak – Microlink User Guide at Page 2 The desired signal – for example, a speaker's voice or the sound from a TV – is picked up by the microphone which is placed near to the source of the sound. The transmitter sends an FM signal to the receiver, which immediately relays this to the hearing instrument for you to hear – crystal clear!

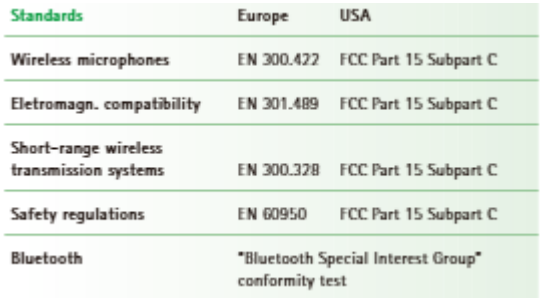
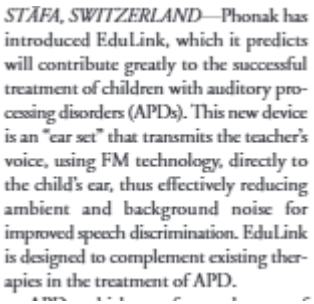
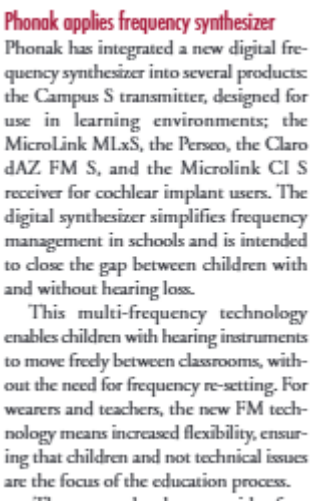
Claims	Prior Art
	Phonak – Annual Report (2003–2004) at p. 12 Within the FM business segment, the Group continued to leverage existing key technology into new applications. With SmartLink, Phonak has achieved another industry first by allowing hearing instrument users to access other communications systems. SmartLink combines a remote control, a transmitter, a directional microphone system and a Bluetooth link to mobile phone all in one single device. Phonak also presented EduLink, an innovative FM device to assist the learning development of children with Auditory Processing Disorder (APD). EduLink was perceived very positively within the medical community and recent classroom tests show favourable results. Phonak – Online Interview re Perseo (April 7, 2003) at Page 9 **Liz Brassine/Phonak:** It is a digital microphone on the CIC. There are several advantages of a digital microphone over analog mics. First, increased dynamic range is provided. Additionally, with a digital mic, the analog-to-digital converter is housed within the microphone itself. This eliminates the wire between the AD-converter and the microphone. The wire is the source for ultra-high frequency interference, such as Bluetooth. Phonak - SmartLink SX - FM Systems Manual at p. 3 SmartLink SX is the first communication device to incorporate • Advanced FM features • Hearing Instrument remote control • A Bluetooth link for cell phone use

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 4 Smart connections to all audio sources Any external audio input or an external microphone signal can be transmitted wirelessly via SmartLink to the hearing instrument, simply by attaching to the smart interface. Phonak - SmartLink SX - FM Systems Manual at p. 10wire  0

Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 11 Smart digital synthesizer technology Digital frequency synthesizer technology, only available with Phonak FM communication systems, enables online frequency synchronization and scanning of the frequencies used. Once a frequency is selected on the SmartLink, it automatically sends a signal to the FM receiver that adjusts the frequency of the receiver accordingly. This automatic synchronization process occurs every time the SmartLink is switched on. In cases where more than one FM transmitter is in use, at lectures or presentations for example, the SmartLink has an exclusive scan function. This seeks out the frequency being used by the other transmitter and sets the receiver frequency accordingly. Phonak - SmartLink SX - FM Systems Manual at p. 14 

Claims	Prior Art
	Phonak - SmartLink SX User Guide at p. 27 • The SmartLink can be directly connected to audio equipment like televisions, CD and MP3 players, and computers with audio outputs. Phonak - SmartLink SX User Guide at p. 32  Phonak - SmartLink SX User Guide at p. 47 

Claims	Prior Art
	Phonak MicroLink MLxS Product Information at Page 1 The MicroLink MLxS is the first miniaturized FM receiver with a digital frequency synthesizer for full multi-frequency flexibility. The receiver features fully automatic, non-contact frequency synchronization with either a WallPilot or the Campus S. The MicroLink MLxS is compatible with all Phonak FM transmitters and the WallPilot, and is recommended for use with this equipment. **MLxS with BTE instrument**  **MLxS with adapter for CI pocket processor**  Phonak – Media Release re Product Offerings in F2003/2004 at Page 3 **Phonak Communications introduces the new MicroLink receivers with frequency synthesizer** Phonak Communications will introduce its new multi frequency MLx S and ML8 S receivers in the next few months. MLx S and ML8 S are the counterpart of Campus S transmitters, already introduced early 2002. Together, they comprise the new synthesized FM system generation, a system that transmits, without wires, speech to the hearing instrument in difficult listening situations. The key technology of the new system is its easy frequency adaptation. Given the extreme design constraints (low voltage, low power, miniature dimensions), the development of this technology represents an unprecedented breakthrough in wireless communication technology.

Claims	Prior Art
	Phonak SmartLink SX Product Information at p. 2  Phonak and Other DHA – Manufacturers News Article at Page 1  Phonak and Other DHA – Manufacturers News Article at Page 2 

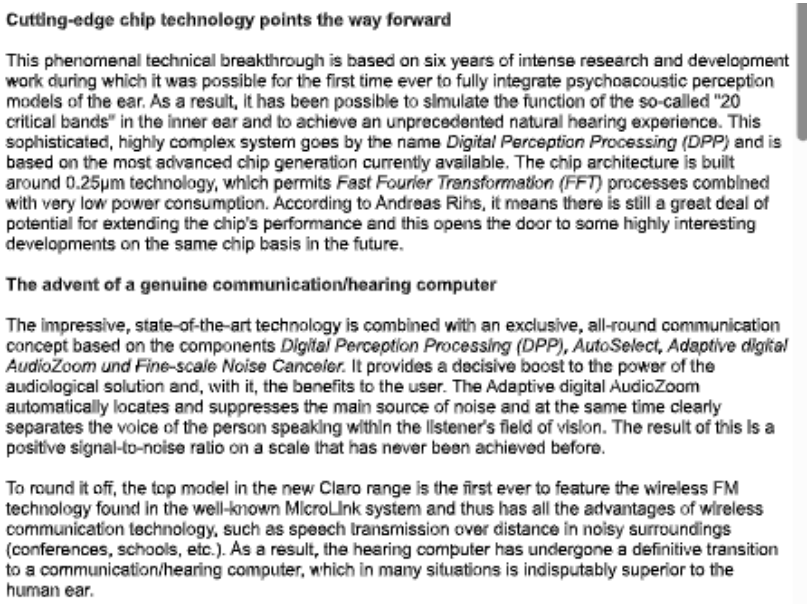
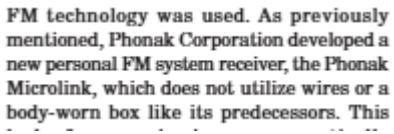
Standards	Europe	USA
Wireless microphones	EN 300.422	FCC Part 15 Subpart C
Eletromagn. compatibility	EN 301.489	FCC Part 15 Subpart C
Short-range wireless transmission systems	EN 300.328	FCC Part 15 Subpart C
Safety regulations	EN 60950	FCC Part 15 Subpart C
Bluetooth	"Bluetooth Special Interest Group" conformity test	

STÄFA, SWITZERLAND—Phonak has introduced EduLink, which it predicts will contribute greatly to the successful treatment of children with auditory processing disorders (APDs). This new device is an "ear set" that transmits the teacher's voice, using FM technology, directly to the child's ear, thus effectively reducing ambient and background noise for improved speech discrimination. EduLink is designed to complement existing therapies in the treatment of APD.

Phonak applies frequency synthesizer

Phonak has integrated a new digital frequency synthesizer into several products: the Campus S transmitter, designed for use in learning environments; the MicroLink MLxS, the Perseo, the Claro dAZ FM S, and the Microlink CI S receiver for cochlear implant users. The digital synthesizer simplifies frequency management in schools and is intended to close the gap between children with and without hearing loss.

This multi-frequency technology enables children with hearing instruments to move freely between classrooms, without the need for frequency re-setting. For wearers and teachers, the new FM technology means increased flexibility, ensuring that children and not technical issues are the focus of the education process.

Claims	Prior Art
	Phonak Claro – World Premiere at Page 1  **Cutting-edge chip technology points the way forward** This phenomenal technical breakthrough is based on six years of intense research and development work during which it was possible for the first time ever to fully integrate psychoacoustic perception models of the ear. As a result, it has been possible to simulate the function of the so-called "20 critical bands" in the inner ear and to achieve an unprecedented natural hearing experience. This sophisticated, highly complex system goes by the name *Digital Perception Processing (DPP)* and is based on the most advanced chip generation currently available. The chip architecture is built around 0.25µm technology, which permits *Fast Fourier Transformation (FFT)* processes combined with very low power consumption. According to Andreas Rihs, it means there is still a great deal of potential for extending the chip's performance and this opens the door to some highly interesting developments on the same chip basis in the future. **The advent of a genuine communication/hearing computer** The impressive, state-of-the-art technology is combined with an exclusive, all-round communication concept based on the components *Digital Perception Processing (DPP), AutoSelect, Adaptive digital AudioZoom und Fine-scale Noise Canceler.* It provides a decisive boost to the power of the audiological solution and, with it, the benefits to the user. The Adaptive digital AudioZoom automatically locates and suppresses the main source of noise and at the same time clearly separates the voice of the person speaking within the listener's field of vision. The result of this is a positive signal-to-noise ratio on a scale that has never been achieved before. To round it off, the top model in the new Claro range is the first ever to feature the wireless FM technology found in the well-known MicroLink system and thus has all the advantages of wireless communication technology, such as speech transmission over distance in noisy surroundings (conferences, schools, etc.). As a result, the hearing computer has undergone a definitive transition to a communication/hearing computer, which in many situations is indisputably superior to the human ear. Lewis – Speech perception in noise: directional microphones (2004) at Page 429  **FM technology was used. As previously mentioned, Phonak Corporation developed a new personal FM system receiver, the Phonak Microlink, which does not utilize wires or a body-worn box like its predecessors. This**

Claims	Prior Art
	Lewis – Speech perception in noise: directional microphones (2004) at Page 431 All subjects were fit with digital Phonak Claro 311 dAZ BTE hearing aids bilaterally. All earmolds had select-a-vent (SAV) venting and #13 or 3 mm horn tubing. In addition to the hearing aids, subjects were fit with Phonak Microlink ML8 FM receivers bilaterally. These FM receivers attach to the bottom of a BTE hearing aid and may be utilized in either the "FM only" mode, which attenuates the hearing aid microphone by 20 dB, or in the FM plus hearing aid mode, which allows for FM input and input of environmental sounds via the hearing aid microphones simultaneously at the same output level. The Phonak TX3 HandyMic FM transmitter served as the FM transmitter. This transmitter has three microphone options: (1) Wide Angle, which picks up sounds arriving from all directions around the transmitter microphone equally; (2) Zoom, which provides reduced amplification to signals arriving from the rear (cardioid); and (3) SuperZoom, which provides reduced amplification to signals originating from the rear and the sides (hypercardioid). The

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 5



Elite

- Claro's digital sound processing with a human perspective (Digital Perception Processing)
- AutoSelect: Excellent speech intelligibility in any listening situation
- Adaptive digital AudioZoom
- Fine-scale Noise Canceler: High-resolution 20-band noise control

Digital

ITE	BTE	Remote controls
Claro 11	Claro 111 dAZ	SoundPilot FM
Claro 21 dAZ	Claro 211 dAZ	WatchPilot Elegance
Claro 22	Claro 311 dAZ	WatchPilot Classic
	Claro 111 dAZ FM	
	Claro 211 dAZ FM	
	Claro 311 dAZ FM	

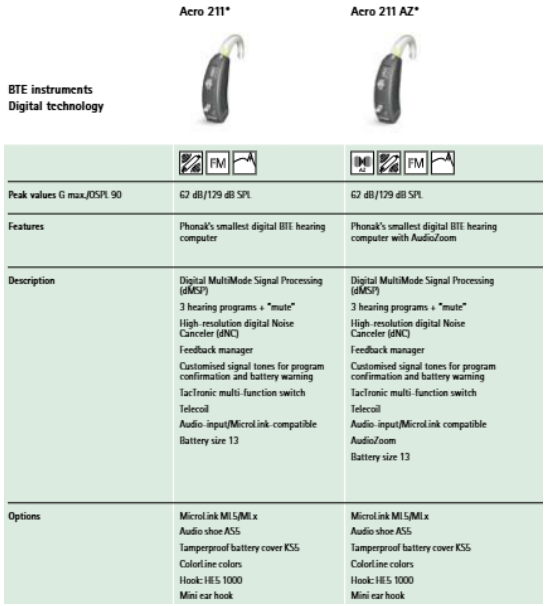
Elite Line features:

- Clear, natural sound with precise, fully automatic volume control
- Significantly improved speech intelligibility even in difficult listening situations
- Design innovations for all system components
- High level of user convenience with automatic hearing program selection
- Innovative high-tech solutions for Claro system control
- State-of-the-art hearing computers for the optimised solution to a wide range of hearing problems
- FM receiver fully integrated in the BTE case (Claro dAZ FM)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 9

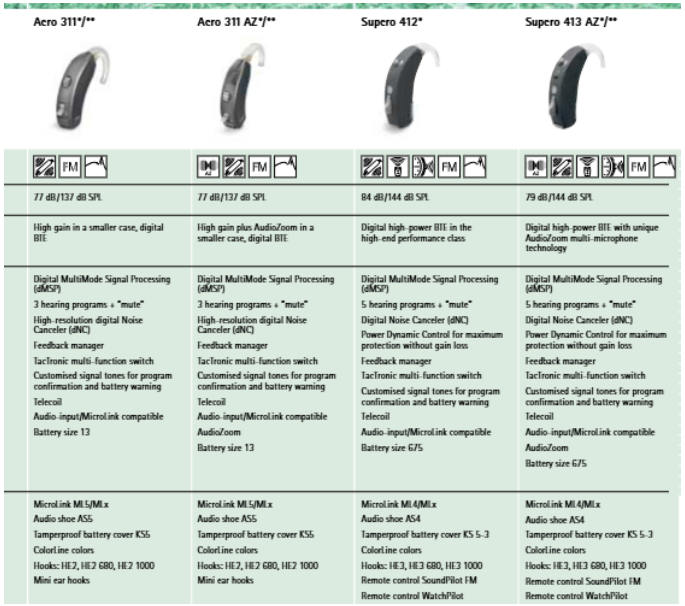
BTE instruments	Claro 111 dAZ Claro 111 dAZ FM	Claro 211 dAZ Claro 211 dAZ FM	Claro 311 dAZ Claro 311 dAZ FM
			
	FM	FM	FM
Peak values G max./OSPL 90	52 dB/126 dB SPL	62 dB/135 dB SPL	72 dB/135 dB SPL
Features	Specially designed for steep high-frequency and mild hearing losses	Digital speech transparency	Digital speech transparency and higher gain
Description	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (111 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (211 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection	Digital Perception Processing (DPP) AutoSelect (AS) Fine-scale Noise Canceler (FNC) Adaptive digital AudioZoom (dAZ) FM module (311 dAZ FM) 5 hearing programs + "mute" Telecoil Battery size 13 Standard: TacTronic program selection switch Wind and weather protection
Options	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)	Remote controls WatchPilot and SoundPilot FM Hooks: HE4, HE4 330, HE4 1000 Tamperproof battery cover ColorLine colors plus F25 (silver), F13 (milky transparent)
Programming	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital, Phonak Digital Ski Slope and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)	Loudness Perception Profile (LPP) with Active Response Unit (ARU) Fitting formulas: Phonak Digital and DSL (i/o)

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 16



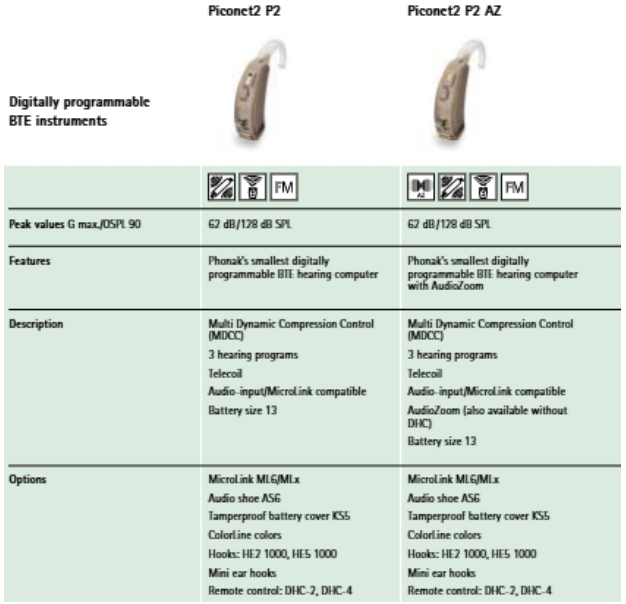
BTE instruments Digital technology	Acro 211*	Acro 211 AZ*
	FM	FM
Peak values G max/OSPL 90	62 dB/129 dB SPL	62 dB/129 dB SPL
Features	Phonak's smallest digital BTE hearing computer	Phonak's smallest digital BTE hearing computer with AudioZoom
Description	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink-compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager Customised signal tones for program confirmation and battery warning TacTronic multi-function switch Telecoil Audio-input/MicroLink compatible AudioZoom Battery size 13
Options	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook	MicroLink MLS/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hook: HE5 1000 Mini ear hook

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 17



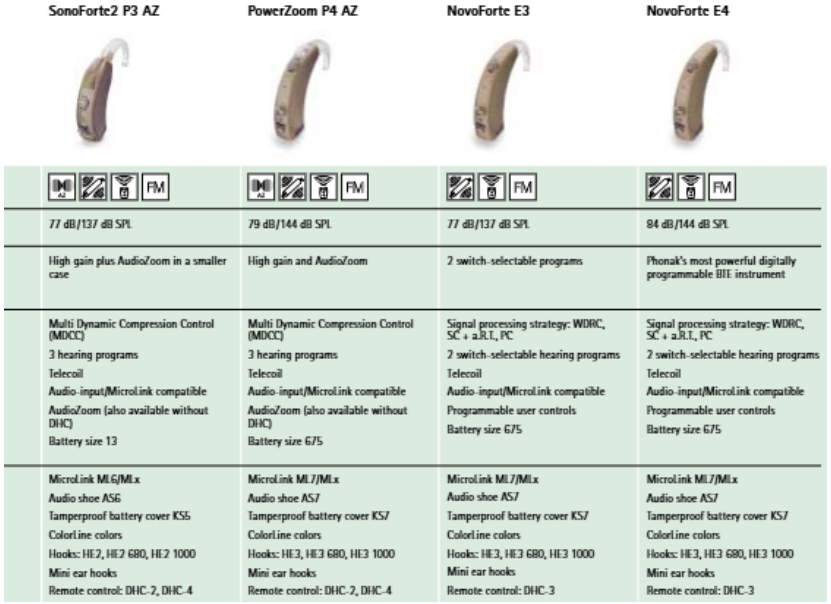
Aero 311*/**	Aero 311 AZ*/**	Supero 412*	Supero 413 AZ*/**
77 dB/137 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL	79 dB/144 dB SPL
High gain in a smaller case, digital BTE	High gain plus Audio/zoom in a smaller case, digital BTE	Digital high-power BTE in the high-end performance class	Digital high-power BTE with unique Audio/zoom multi-microphone technology
Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 13	Digital MultiMode Signal Processing (dMSP) 3 hearing programs + "mute" High-resolution digital Noise Canceler (dNC) Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 13	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Digital MultiMode Signal Processing (dMSP) 5 hearing programs + "mute" Digital Noise Canceler (dNC) Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Audio/zoom Battery size 675
MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Remote control SoundPilot FM Remote control WatchPilot

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 18



Digitally programmable BTE instruments	Piconet2 P2	Piconet2 P2 AZ
	FM	FM
Peak values G max/OSPL 90	62 dB/128 dB SPL	62 dB/128 dB SPL
Features	Phonak's smallest digitally programmable BTE hearing computer	Phonak's smallest digitally programmable BTE hearing computer with AudioZoom
Description	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13
Options	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS5 ColorLine colors Hooks: HE2 1000, HE5 1000 Mini ear hooks Remote control: DHC-2, DHC-4

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 19

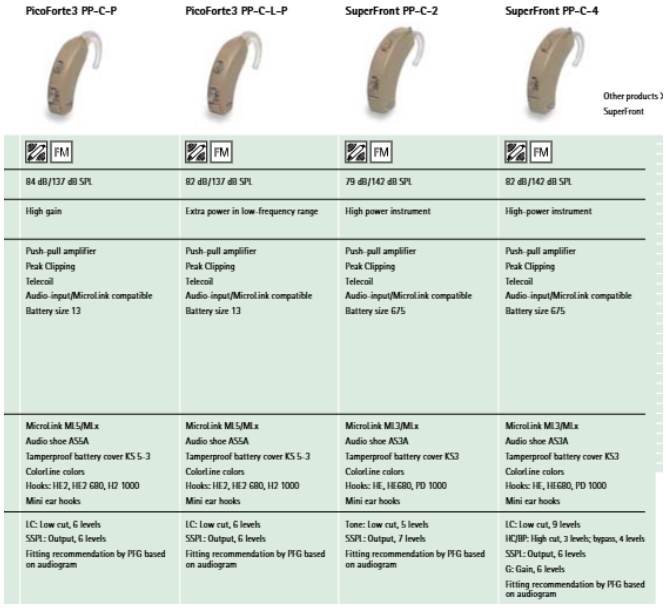


SonoForte2 P3 AZ	PowerZoom P4 AZ	NovoForte E3	NovoForte E4
FM	FM	FM	FM
77 dB/137 dB SPL	79 dB/144 dB SPL	77 dB/137 dB SPL	84 dB/144 dB SPL
High gain plus AudioZoom in a smaller case	High gain and AudioZoom	2 switch-selectable programs	Phonak's most powerful digitally programmable BTE instrument
Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 13	Multi Dynamic Compression Control (MDCC) 3 hearing programs Telecoil Audio-input/MicroLink compatible AudioZoom (also available without DHC) Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675	Signal processing strategy: WDRC, SC + a.R.T., PC 2 switch-selectable hearing programs Telecoil Audio-input/MicroLink compatible Programmable user controls Battery size 675
MicroLink ML6/MLx Audio shoe AS6 Tamperproof battery cover KS6 ColorLine colors Hooks: HE2, HE2 680, HE2 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-2, DHC-4	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3	MicroLink ML7/MLx Audio shoe AS7 Tamperproof battery cover KS7 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks Remote control: DHC-3

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 24

BTE instruments	Supero 411*	PicoForte3 SC-D	PicoForte3 PP-SC
			
	FM	FM	FM
Peak values G max./OSPL 90	79 dB/142 dB SPL	64 dB/127 dB SPL	78 dB/133 dB SPL
Features	Digital high-power BTE	Directional microphone, switchable	High power
Description	dLimiting signal processing strategy 1 hearing program + "mute" Power Dynamic Control for maximum protection without gain loss Feedback manager TacTronic multi-function switch Customised signal tones for program confirmation and battery warning Telecoil Audio-input/MicroLink compatible Battery size 675	Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13	Super Compression + a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 13
Options	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE3, HE3 680, H3 1000	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks
Programming	Programmable with PC (IBM-compatible) and HI-PRO programming interface PFG Software, version 8.0 or above	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 25



PicoForte3 PP-C-P	PicoForte3 PP-C-L-P	SuperFront PP-C-2	SuperFront PP-C-4
			Other products) Superl·ront
FM	FM	FM	FM
84 dB/137 dB SPL	82 dB/137 dB SPL	79 dB/142 dB SPL	82 dB/142 dB SPL
High gain	Extra power in low-frequency range	High power instrument	High-power instrument
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 13	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML5/MLx Audio shoe AS5A Tamperproof battery cover KS 5-3 ColorLine colors Hooks: HE2, HE2 680, H2 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3A Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks
LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 6 levels SSPL: Output, 6 levels Fitting recommendation by PFG based on audiogram	Tone: Low cut, 5 levels SSPL: Output, 7 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels HC/BP: High cut, 3 levels; bypass, 4 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram

IngenioSpec EX2021
Apple v. IngenioSpec, IPR2026-00331
Page 481

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 26

	SuperFront PP-C-L-4 SuperFront PP-C-L-4+	SuperFront PP-SC	Classica AGC
BTE instruments			
	FM	FM	FM
Peak values G max./OSPL 90	82 dB/142 dB SPL (PP-C-L-4) 81 dB/142 dB SPL (PP-C-L-4+)	82 dB/140 dB SPL	65 dB/133 dB SPL
Features	High-power low-freq. range (PP-C-L-4) Super low-freq. instrument (PP-C-L-4+)	High-power	AGC-O with a.R.T.
Description	Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Broadband response Telecoil Audio-input/MicroLink compatible Battery size 675
Options	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML3/MLx Audio shoe AS3 Tamperproof battery cover KS3 ColorLine colors Hooks: HE, HE680, PD 1000 Mini ear hooks	MicroLink ML4/MLx Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
Programming	LC: Low cut, 9 levels HC: High cut, 6 levels SSPL: Output, 6 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 9 levels BP/HC: Bypass, 6 levels; high cut, 3 levels SSPL: Output, 9 levels G: Gain, 6 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 27

Classica C-D	Classica PP-SC	Classica PP-C-P	Classica PP-C-L-P
			
FM	FM	FM	FM
70 dB/139 dB SPL	80 dB/137 dB SPL	84 dB/144 dB SPL	80 dB/144 dB SPL
Broadband BTE with directional microphone	High-power	High gain	Extra power in low-frequency range
Push-pull amplifier Peak Clipping Telecoil Audio-input/MicroLink-compatible Battery size 675	Push-pull amplifier Super Compression with a.R.T. Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675	Push-pull amplifier Peak Clipping Broadband Telecoil Audio-input/MicroLink compatible Battery size 675
MicroLink MI 4/MI x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink MI 4/MI x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink MI 4/MI x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks	MicroLink MI 4/MI x Audio shoe AS4 Tamperproof battery cover KS4 ColorLine colors Hooks: HE3, HE3 680, HE3 1000 Mini ear hooks
LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram	LC: Low cut, 5 levels SSPL: Output, 5 levels Fitting recommendation by PFG based on audiogram

Claims	Prior Art
	Phonak – Collections 2002 Product List at Page 30

MLx, the most flexible FM receiver ever made, can be used not only with Phonak BTEs but also with most third-party hearing instruments available on the market today. This is possible thanks to the standard Euro connector which simply plugs into the audio shoe of the hearing instrument.

Once again, Phonak demonstrates its commitment to innovative developments that serve hearing-impaired adults and children.

MLx compatible BTE makes:

- Argosy
- Audifon
- Beltone
- Danavox
- GN ReSound
- Hansaton
- Interton
- Oticon
- Philips
- Phonak
- Rexton
- Siemens
- Sonic Innovations
- Unitron
- Widex









Detailed, up-to-date compatibility lists of Phonak's MLx MicroLink receiver are available online at:

www.phonak.com/mlx

Phonak – Claro 211 Product Information at Page 1

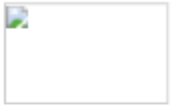
FM features

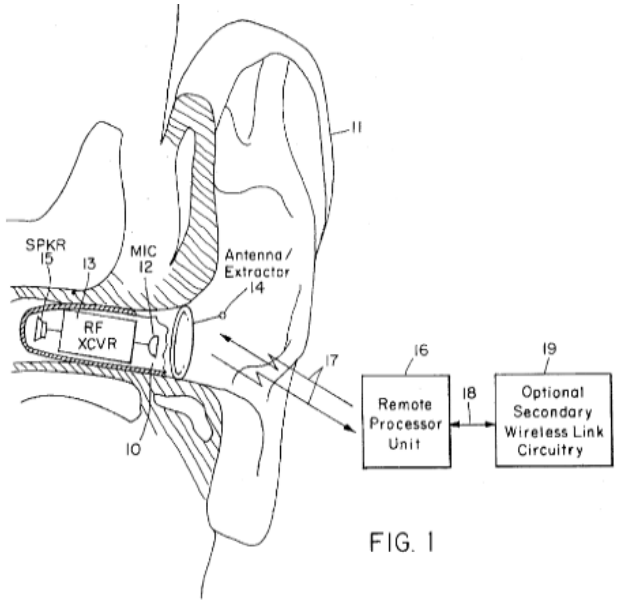
- World's smallest FM microchip
- Compatible with all Phonak FM transmitters
- Cordless TV and Radio link up
- FM performance corresponding to audiological fitting
- Soft Squelch system
- Built in antenna
- Band H (173–185 MHz); band J (186–201 MHz); band N (207–217 MHz)

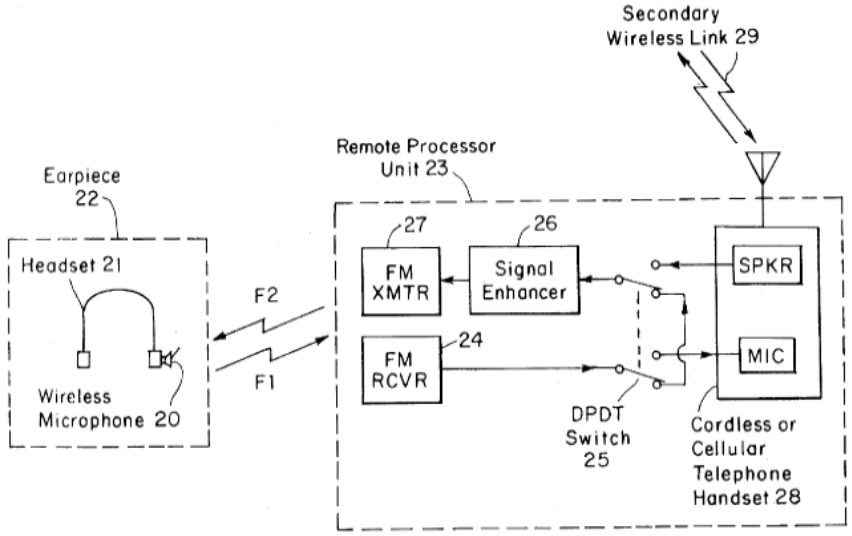
Phonak – Claro 211 Product Information at Page 1

Processing features

- Digital Perception Processing (DPP) in 20 critical bands
- Gain and output controlled by DPP
- 3 hearing programs (+ mute mode):
  - QuietAdapt for hearing in relative quiet
  - NoiseAdapt for hearing speech in noise
  - Program 3 telephone default (other choices available)
- FM programs: FM and FM+M
- AutoSelect: automatic switching between QuietAdapt and NoiseAdapt
- Claro's noise suppression technologies
  - Adaptive digital AudioZoom (dAZ)
  - Fine-scale Noise Canceler in 20 bands
  - Soft squelch

Claims	Prior Art
	Phonak – Claro 211 Product Information at Page 2 **Principle of FM operation** The transmitter unit is worn by the person speaking, placed in front of the sound source or even connected directly to TV or HiFi. The transmitter unit collects the desired signal and transmits it via radio waves directly to the FM module integrated in the hearing instrument. The incoming FM signal received by the hearing instrument is processed either in accordance with the audiological setting of the instrument, providing consistent sound quality or according to an individual FM program. Phonak – FM Transmitters including Campus S and TelCom at Page 1 TelCom  TelCom (click to enlarge) Phonak's TelCom is the first stationary FM transmitter that **allows optimized television, telephone, and audio system usage**. While the user watches TV or listens to music, TelCom assures that the telephone will be heard when it rings. When the call is answered, TelCom automatically mutes the audio signal and transmits the caller's voice clearly and crisply. This does not affect the volume of the loudspeakers for others in the room. When the handset is replaced, the audio signal transmitted to the hearing instruments is turned back up to the previous volume setting. The TelCom has an operating range of 7 to 15 meters (23 to 50 feet). **For television, telephone, CD player, radio, personal computer and other equipment with audio output**. Oticon – DigiFocus White Paper (1996) at Page 11 Many people find it uncomfortable to be wired to a computer. This "constricted" feeling can be avoided by using the OtiSet programming belt, which is worn on the shoulder. The programming belt provides cordless infrared communication between the PC and the hearing instrument while in the ear. Alternatively, you may use the HI-PRO interface for a wired solution. OtiSet offers a choice of three different views of the fitting. The Audiological View provides a visual guide to instrument performance in four typical sound environments. The Technical View illustrates traditional technical information such as coupler data and insertion gain at different input levels. With the Binaural View you are able to examine settings for both right and left instruments at the same time. Anderson '783 at Abstract A hearing aid or audio communication system includes an earpiece (10) that can be hidden in the ear canal, and which communicates wirelessly with a remote processor unit, or RPU (16), that enhances audio signals and can be concealed under clothing. Sounds from the environment are picked up by a microphone

Claims	Prior Art
	(12) in the earpiece and sent with other information over a two-way wireless link (17) to the RPU (16). The wireless link (17) uses microwaves for component miniaturization. Furthermore, use of radar technology to implement the wireless link (17), with an RPU (16) interrogator and earpiece (10) transponder, reduces earpiece size and power, as no microwave oscillator is needed in the earpiece (10). Optional secondary wireless link circuitry (19) can be used between the RPU (16) and a cellular telephone system or other sources of information. Electronic voice recognition and response can control system operation. Anderson '783 at FIG. 1  FIG. 1

Claims	Prior Art
	Anderson ’783 at FIG. 2  FIG. 2 Anderson ’783 at 1:5–18 The present invention is directed to hearing aids, and in particular to a hearing aid having an earpiece housing worn in or at the ear and a remote processor unit (RPU) worn by or located near the user that wirelessly receives signals from and transmits signals to the earpiece. The present invention is also directed to the use of hearing aid systems as wireless communication devices for hands-free cellular telephone and mobile radio communication "handsets," covert operation and control of hearing aids, hearing protection and noise cancellation simultaneous with binaural hearing aid functions, hearing test equipment, location of misplaced hearing aid system components and wireless cochlear implants. Anderson ’783 at 1:50–65 The present invention uses an audio path that is different from the audio path used by devices in the prior art. The present invention uses a primary two-way wireless link (not a one-way link as in the prior art) between an earpiece worn in or at the user's ear and an RPU worn by or located near the user. In the present invention, audio signals from the environment are picked up by a microphone in the earpiece (not a hand-held microphone as used by wireless hearing aids in the prior art) and transmitted over the primary two-way wireless link to the RPU (instead of going to processor electronics contained in the earpiece, as

Claims	Prior Art
	in basic hearing aids known in the prior art), where the audio signals are enhanced according to the user's needs before transmission over the primary wireless link to the earpiece. Signal processing is performed in the RPU rather than the earpiece to take advantage of relaxed size and power constraints. Anderson ’783 at 2:20–39 It is an object of the present invention to provide a new and useful auditory aid for hearing-impaired persons (i.e., those having certain residual hearing) by removing audio signal enhancement functions from the earpiece and placing them in an RPU. Use of an RPU provides several advantages over systems that attempt to place all system capabilities within the earpiece. The RPU approach allows a simple earpiece design comprising a miniature low-power wireless transceiver, microphone and speaker. Note that the speaker is also known in the hearing aid art as a "receiver," but the term "speaker" is used here to avoid confusion (similarly, the term "talker," not "speaker," is used to describe a human producing vocal sounds). The resulting earpiece is extremely small, can be hidden from view in the ear canal if desired, and allows complete freedom of movement when a primary two-way wireless link to the RPU is used. Processing for all major system capabilities, such as amplification and other forms of signal enhancement, takes place in the RPU where size and power constraints are relaxed, leading to a cost-effective design. Anderson ’783 at 2:40–59 Another object of this invention is to provide supplemental audio information (e.g. a verbal warning from the RPU that an earpiece battery is low) and communication services (e.g., cellular telephone and paging services) to the user via the hearing aid system. Such services are accessed in a manner that can be made imperceptible to a casual observer if desired. The RPU (which contains a digital signal processor or other computer) acts as one source of information, e.g., by using a synthesized voice message to provide the time of day, and the RPU can also be used to access a secondary wireless link to the general subscriber telephone network or voice paging services. User control of hearing aid parameters and requests for information are accomplished using pushbuttons located on the RPU (including pushbuttons suitable for data entry in a covert manner) or voice recognition. Many sensors and peripheral devices can reside in or be attached to the RPU by wired or wireless means, and can provide a variety of information for different applications (e.g., heart pulse rate) as audio in the user's earpiece. Anderson ’783 at 2:60–3:5 A further object of this invention is to provide some degree of protection for the residual hearing capability of a hearing-impaired user (or a non-impaired user who wishes to avoid impairment) in a wireless hearing aid system having the simultaneous capabilities of noise cancellation and binaural

Claims	Prior Art
	processing (e.g., directionality). Other objects of the invention are to provide a convenient means for testing a user's hearing capability without the need for additional equipment, to assist users in the location of misplaced hearing aid system components, and to provide profoundly deaf cochlear implant patients with a wireless system allowing improved freedom of movement compared to existing wire-connected systems. Anderson '783 at 3:52–4:14 One preferred embodiment of the invention herein described is shown in block diagram form in FIG. 1. In this figure, the earpiece 10 shown worn in the ear 11 uses a standard completely-in-the-canal (CIC) housing, well known in the art, although many other earpiece housing types known in the art (e.g., behind-the-ear, or BTE) can be used. The earpiece shown in FIG. 1 comprises a microphone 12, RF transceiver 13 with an antenna that doubles as an earpiece extractor 14, and speaker 15. Although the dual-use antenna/extractor arrangement shown in FIG. 1 is often desirable, a separate antenna or antennae can be used in conjunction with many different types of earpiece extractors as appropriate for a particular application. Antennae and extractors can be disguised as ear hair or jewelry (e.g., earrings), or conductive filaments can be permanently implanted in the ear cartilage to act as antennae using a process similar to ear piercing. The earpiece 10 communicates with a remote processor unit (RPU) 16 via a primary two-way RF link 17, although many other wireless link media (e.g., ultrasonic or infrared) may be used instead. The RPU is typically worn under clothing (e.g., carried in a pocket or purse), but may also be worn in plain sight (e.g., on a belt) if desired. The RPU 16 may be connected (via wired or wireless means 18) to optional secondary wireless link circuitry 19 that allows wireless communication between the RPU and other sources of information (e.g., the general subscriber telephone network) via a secondary wireless link. Note that the optional secondary wireless link circuitry 19 may or may not be contained within the RPU case 70 of FIG. 7. Anderson '783 at 5:3–21 During normal operation, speech signals from a nearby talker (and other signals in the ambient audio environment) are picked up by the earpiece wireless microphone 20 and transmitted to the RPU FM receiver 24. The RPU FM receiver 24 output level may be adjusted using the RPU FM receiver 24 volume control. The resulting electrical waveform representing signals from the nearby talker travels through the DPDT switch 25, which is set to the lower position as shown in FIG. 2, to the signal enhancer 26. The signal enhancer 26 may be, for example, a voice changer device that varies the pitch of a received speech signal according to the setting of pushbutton controls located on the RPU signal enhancer 26. The speech signal's pitch can then be raised or lowered as desired to help compensate for a user's hearing loss relative

Claims	Prior Art
	to a particular talker's voice characteristics. The modified speech signal travels from the RPU signal enhancer 26 to the RPU FM transmitter 27, and finally to the earpiece headset 21 receiver where the signal is converted to acoustic waves heard by the user. Anderson ’783 at 6:8–25 The user's voice (and other signals in the ambient audio environment) is picked up by the earpiece wireless microphone 20, transmitted to the RPU 23 FM receiver 24, sent through the DPDT switch 25 to the microphone input of the handset 28 (the handset's microphone transducer may be removed) and transmitted on the secondary wireless link 29 to, e.g., the general subscriber telephone network. Signals from the secondary wireless link 29 are received by the handset 28, sent from the speaker output of the handset 28 (the handset's speaker may be removed) via the DPDT switch 25 to the signal enhancer 26, FM transmitter 27 and finally to the earpiece headset 21 receiver where the signals are converted to acoustic waves heard by the user. Note that the RPU 23 may also include straight-forward connections to a "personal digital assistant" computer, or voice-operated devices connected to the FM receiver 24 output and not shown in FIG. 2. Anderson ’783 at 21:50–62 In the simplest case, the ASIC 845 reads data from the A/D 880, performs a multiplication of the data by a constant (read from the default parameter storage 870) to provide amplification, compares the result with maximum allowable values (read from the default parameter storage 870) and writes the result to the D/A 850. The ASIC 845 may perform other processing (e.g., adaptive volume control) as well. Although none of the remote processor and communication features are available in stand-alone mode, which uses pre-selected hearing aid parameter settings from the default parameter storage 870 in the earpiece, the earpiece provides a useful amplified audio path from the microphone to the speaker. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other

Claims	Prior Art
	information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948. Anderson ’783 at 24:26–32 Sensor measurements can be reported to the user by a voice synthesized in the RPU DSP 948 using any of a variety of synthesis techniques well known in the art, including playback of pre-recorded digitized speech. The voice signal produced within the RPU DSP 948 is then added to the processed audio signals already being sent from the RPU to the earpiece. Anderson ’783 at 24:45–47 Solicited information: Solicited information is provided by the system (e.g., via the earpiece speaker 15)

Claims	Prior Art
	in response to user requests (e.g., via an RPU keyboard 946, Anderson ’783 at 24:47–66 Morse code keys 956 or voice recognition capabilities implemented using the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. In this respect, the system acts as a "virtual handset" communication aid that replaces a conventional telephone handset in many applications (although users may additionally communicate with a conventional handset or speakerphone in the usual fashion). To accomplish the virtual handset function, the RPU 16 is equipped with optional secondary two-way wireless link circuitry 19 to communicate with a telephone or other communication network. The secondary wireless link is separate from, and does not interfere with, the primary wireless link 17 between the earpiece 10 and RPU 16. In applications where relatively high-power transmissions take place on the secondary link and the secondary link transmitter is integrated into the RPU, the RPU can be placed on a user's wrist or ankle away from vital organs to reduce any potential health risk relative to conventional wireless handsets. Anderson ’783 at 25:15–27 To obtain solicited information, user requests are processed in a variety of means well known in the art, including inputs via one or more pushbuttons on the RPU and voice recognition computer programs implemented in the RPU electronics. Voice recognition can be used almost exclusively, if desired, thereby eliminating nearly all RPU pushbutton controls. Voice commands recognized by the RPU can cause the RPU to initiate actions such as changing hearing aid parameter settings (e.g., volume), or activate appliances and environmental controls (e.g., aids for the physically handicapped) in a distributed wireless network via the secondary wireless link. Anderson ’783 at 25:50–26:5 Sensors and peripheral devices: Many sensors and peripheral devices 950 can reside in or be attached to the RPU by wired or wireless means, and controlled by the user (via inputs to the RPU DSP 948 through the keyboard 946, Morse code keys 956 or voice recognition) to provide both solicited and unsolicited information. Music can be provided from pre-stored information including tape, compact disc, over-the-telephone services via the secondary wireless link, or a commercial broadcast radio receiver. User location information can be provided to the RPU from special-purpose transmitters at specific locations in nursing home applications. Self-contained user-carried inertial navigation packages (with solid-state gyroscopes, accelerometers and a compass) or a Global Positioning System receiver can also be used. Sensors for a foot soldier RPU application may include many types of detectors such as RF detectors

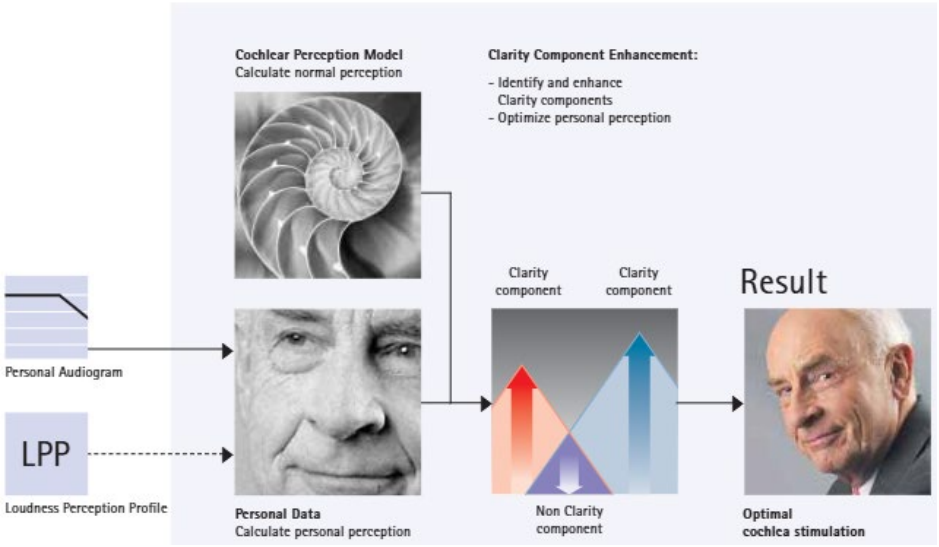
Claims	Prior Art
	(e.g., radar detection capability), nuclear/biological/chemical weapon detectors, infrared or laser detectors and metal detectors. Appropriate verbal warnings or distinctive alerting tones generated by the RPU DSP 948 are sent from the RPU 16 to the earpiece 10 and ultimately received as audio signals by the user. Enzmann '730 at 1:7–11 This invention relates to systems and methods for providing a hearing aid capable of communicating with a wireless phone. More particularly, this invention provides a hearing aid capable of simultaneously or alternatively performing the functions of an ear bud for the wireless phone. Enzmann '730 at 2:17–23 Therefore, this invention provides systems and methods for adapting a hearing aid to operate as an ear bud device in addition to providing conventional hearing aid functions. Using this method, this invention is able to address many of the issues hearing impaired wireless phone users encounter while addressing the inherent problems associated with the conventional solutions. Enzmann '730 at 3:55–57 In addition, the portability of the wireless phone is increased because the user's hearing device can now act as a wireless ear bud for the wireless phone. Enzmann '730 at 4:14–16 In the second operating mode, the TX/RX unit 260 transmits and receives communications from the wireless phone 270. Enzmann '730 at 4:43–50 The connection shown between the TX/RX unit 260 and the wireless phone 270 is preferably a wireless digital connection, such as Bluetooth, etc. However, it should be appreciated that the connection between the TX/RX unit 260 and the wireless phone 270 can be any kind of wireless or wired connection, and may be digital or analog, infrared (IR) or transmitted radio frequencies (RF) without, departing from the scope of the invention. Enzmann '730 at 4:56–62 In the ear bud mode of operation, as mentioned above, the switch 231 is activated (i.e., closed), providing a continuous signal path for signals coming from the op amp 222 to be passed to the TX/RX unit 260

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	along signal line 226. Additionally, the TX/RX unit 260 can pass data to the op amp 223 along signal line 228 though the signal processing function 225 and signal line 228a. Enzmann ’730 at 4:63–5:5 Signals received at the microphone 205 are converted by the A/D converter 210, manipulated by the signal-processing function 221 which outputs 226a is then passed to the op amp 222. The output 226 and 226b of the op amp 222 are electronically connected to the TX/RX unit 260 and to the op amp 223. The output 226 connected to the TX/RX unit 260 is then output to the wireless phone 270. The signal from the TX/RX unit 260 is used by the wireless phone 270 as if the user was speaking into the microphone of the wireless phone. Enzmann ’730 at 5:13–19 The signals received at the microphone 205 are passed to the user as background noise and must be passed to the wireless phone 270 as if the user was speaking directly into the wireless phone 270. Remotely transmitting this signal to the wireless phone from the aided-ear bud 200 also allows the user to be located in a remote location away from the wireless phone 270. Enzmann ’730 at 5:20–30 The signal 226b passed to the op amp 223 is representative of ambient noise in the aided ear bud 200. This signal 226b is typically reduced in volume and combined with the input from the signal-processing function 225 to produce a mixed signal. The signal processing function 225 processes signal 228 from the TX/RX unit 260 and provides a second processing step so that the timing of the incoming signals from the microphone and the wireless phone are in sync and can be effectively combined to create a true representation of sound to the user. The op amp 223 outputs the mixed signal to the D/A converter 240, and finally to the speaker 250. Enzmann ’730 at 6:12–35 The amplification chart 325 contains a series of pre-stored dynamic frequency maps based on the user's hearing characteristics, the speaker characteristics and the ambient noise near the aided ear bud 200. The digital logic 320 will receive the transmitted signal from the TX/RX unit 260 and process those signals to an optimal form based on the user's hearing, speaker and ambient noise characteristics. The arrangement of the logic devices within the digital logic 320 will serve to produce each of the characteristics (wave shape) of the amplitude chart 325. The user's hearing characteristics remain static unless the aided ear bud

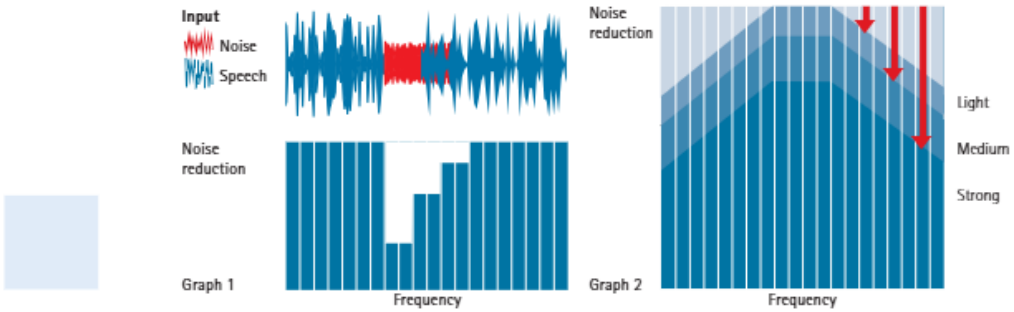
Claims	Prior Art
	200 is reprogrammed to compensate for additional hearing loss or increase and different users. For example, as shown in FIG. 5, a mapping of a user's hearing characteristic (denoted by the "••" line) illustrates that the user lacks significant hearing capability in higher frequencies. The desired hearing characteristic is denoted by the "-••" line. This desired hearing characteristic depicts or represents how a non-hearing impaired person's hearing characteristics may be. As known to those skilled in the art, close representation of this hearing characteristic is achievable by adding a complementary characteristic to the user's hearing characteristic to produce a result that is very close to the desired characteristic. Each of the characteristics is corrected using the process shown in FIG. 5. Jannard '526 at Abstract A wearable audio interface includes a support for positioning the plurality of speakers juxtaposed to and spaced from the ears of a wearer. The audio device can include wireless networking electronics so as to allow the device to interact with other wireless network devices. Jannard '526 at 1:26–33 There are numerous situations in which it is convenient and preferable to mount audio output devices so that they can be worn on the head of a user. Such devices can be used for portable entertainment, personal communications, and the like. For example, these devices could be used in conjunction with cellular telephones, cordless telephones, radios, tape players, MP3 players, portable video systems, hand-held computers and laptop computers. Jannard '526 at 2:10–13 The system also comprises source electronics electrically coupled with the receiver electronics and configured to wirelessly transmit information to the receiver electronics. Jannard '526 at 2:14–22 In one implementation, the source electronics are configured to wirelessly receive the information that the source electronics transmits to the receiver electronics. In a further implementation, the source electronics comprises a satellite. In a further implementation, the satellite comprises a source of global positioning to determine the position of the wearer. In another implementation, the source electronics comprises a source of music.

Claims	Prior Art
	Jannard ’526 at 19:41–45 As illustrated in FIG. 8, the audio device 10C can be worn on the head 18 of a user U. Preferably, the audio device 10C is configured to provide one or two-way wireless communication with a source device, or the source device can be incorporated into the audio device 10C. Jannard ’526 at 19:48–54 The user U can carry a “body borne” source device B such as, for example, but without limitation, a cellular phone, an MP3 player, a “two-way” radio, a palmtop computer, or a laptop computer. As such, the user U can use the audio device 10C to receive and listen to audio signals from the source device B, and/or transmit audio signals to the source device B. Jannard ’526 at 25:12–21 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the audio device 10C is advantageously adapted to support any of a variety of portable electronic circuitry or devices which have previously been difficult to incorporate into conventional headsets due to bulk, weight or other considerations. For example, but without limitation, the electronics are digital or other storage devices and retrieval circuitry such as for retrieving music or other information from MP3 format memory or other memory devices. Jannard ’526 at 28:4–37 With reference to FIGS. 17 and 18, a communication protocol between the audio device S, B and the transceiver 114 is described. In this embodiment, the transceiver 114 is configured for one-way communication. The transceiver includes a receiver and decoder 202 and a digital-to-audio converter 204. As noted above with reference to FIG. 8, the audio device S, B can be any one of a number of different audio devices. For example, but without limitation, the audio device S, B can be a personal audio player such as a tape player, a CD player, a DVD player, an MP3 player, and the like. Alternatively, where the transceiver 114 is configured only to transmit a signal, the audio device S, B can be, for example, but without limitation, an audio recording device, a palm top computer, a laptop computer, a cell phone, and the like. For purposes of illustration, the audio device S, B will be configured only to transmit a signal to the transceiver 114. Thus, in this embodiment, the audio device S, B includes an MP3 player 206 and an encoder and transmitter 208. An antenna 210 is illustrated schematically and is connected to the encoder and transmitter 208. As an illustrative example, the MP3 player 206 outputs a signal at 128 kbps (NRZ data). However, other data rates can be used. The encoder and transmitter 208 is configured to encode the 128 kbps signal from the MP3 player and to transmit it through the antenna 210. For example, the encoder

Claims	Prior Art
	and transmitter 208 can be configured to transmit the encoded signal on a carrier signal centered on 49 MHz. The receiver and decoder 202 can be configured to receive the carrier signal of 49 MHz through the antenna 118, decode the digital signal, and transmit the digital signal to the digital-to-audio converter 204. The digital-to-audio converter 204 can be connected to the speakers 14,16 and thereby provide an audio output that is audible to the user. Jannard '526 at 30:40–52 The headset profile is designed to be used for interfacing a headset having one earphone, a microphone, and a transceiver worn by the wearer, for example, on a belt clip, with a cordless phone through a wireless connection. According to the headset profile, certain commands can be issued from a headset, such as the audio devices 10, 10A, 10A′, 10B, 10C, 10D, and 10E, using an AT command protocol. In such a protocol, text commands must be input to the BLUETOOTH™ device, which the BLUETOOTH™ device then transmits wirelessly to a synchronized BLUETOOTH™ enabled device. Such commands include, for example, but without limitation, initiating a call, terminating a call, and redialing a previously dialed number. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
59. A head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output signals result from amplifying or attenuating different frequency bands differently according to at least the predetermined hearing characteristics associated with the user.	DHA systems, alone or in view of Anderson, Jannard, or Rader, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 17, wherein the customized audio output signals result from amplifying or attenuating different frequency bands differently according to at least the predetermined hearing characteristics associated with the user."

Claims	Prior Art
	Phonak – Perseo Brochure at Page 13 Enhanced sound quality and Speech-to-Noise Ratio **Clarity Component Enhancement – DPP²** Speech clarity depends on both its acoustic properties and the listener's auditory perception. Conventional spectral enhancement systems consider only the acoustics of sound, not how it will be perceived by the listener. Perseo goes well beyond this with **DPP²**, the new generation of Digital Perception Processing. **DPP²** with **Clarity Component Enhancement** is individualized for each wearer. It analyzes the acoustic and the psychoacoustic characteristics of incoming sound in 20 critical bands to identify accurately the components that contribute to clarity. It then selectively amplifies them to optimize clarity for the wearer. Phonak – Perseo Brochure at Page 13 Digital Perception Processing² (DPP²) 

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	Phonak – Perseo Brochure at Page 14 Enhanced comfort in all situations **Fine-scale Noise Canceler in 20 channels** Finely scaled frequency resolution in 20 channels gives the **Fine-scale Noise Canceler (FNC)** the most precise background noise reduction capability. Noise reduction depends on the relative amounts of speech and noise in each channel (graph 1) and the importance of each channel for speech understanding. The **FNC** in Perseo ensures maximum comfort and ease of listening. Annoying background sounds are reduced for comfort while the important sounds of speech are preserved for stress-free listening. The maximum amount of noise reduction can be fitter-programmed to match the individual wearer's tastes and requirements. Thanks to the **dual-base automatic**, separate settings can be made for relatively quiet and noisy situations, ensuring optimal comfort in multiple situations (graph 2).  Phonak - SmartLink SX - FM Systems Manual at p. 4 Smart signal processing For the first time in the industry, SmartLink combines all of the verified advantages of digital signal processing strategies with the proven Phonak miniaturized MicroLink technology. SmartLink – the gateway to crystal-clear sound quality Phonak - SmartLink SX - FM Systems Manual at p. 6 Advanced DSP processing package: • Digital SuperZoom² • Fine-scale Noise Canceler • Digital voice enhancement

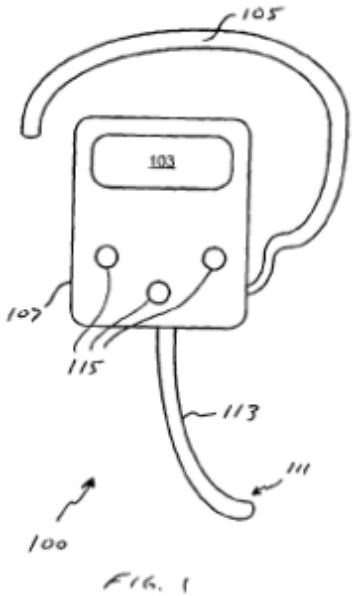
Claims	Prior Art
	Phonak - SmartLink SX - FM Systems Manual at p. 9 The SmartLink three-phase signal recognition and audio processing strategy is the key to creating natural, crystal-clear sound. In the first phase, the focus of reception is defined through the choice of microphone settings. Secondly, the noise cancellation eliminates disturbing noises detected within the defined beam. Finally, voices within the defined beam are enhanced through digital voice enhancement technology. Phonak - SmartLink SX - FM Systems Manual at p. 9 Digital voice enhancement By analyzing the acoustic characteristics of the incoming sound, the digital voice enhancement feature optimizes the perceived loudness of the transmitted FM signal. Anderson ’783 at 2:20–39 It is an object of the present invention to provide a new and useful auditory aid for hearing-impaired persons (i.e., those having certain residual hearing) by removing audio signal enhancement functions from the earpiece and placing them in an RPU. Use of an RPU provides several advantages over systems that attempt to place all system capabilities within the earpiece. The RPU approach allows a simple earpiece design comprising a miniature low-power wireless transceiver, microphone and speaker. Note that the speaker is also known in the hearing aid art as a "receiver," but the term "speaker" is used here to avoid confusion (similarly, the term "talker," not "speaker," is used to describe a human producing vocal sounds). The resulting earpiece is extremely small, can be hidden from view in the ear canal if desired, and allows complete freedom of movement when a primary two-way wireless link to the RPU is used. Processing for all major system capabilities, such as amplification and other forms of signal enhancement, takes place in the RPU where size and power constraints are relaxed, leading to a cost-effective design. Anderson ’783 at 22:61–23:35 Details of normal system operation: During normal operation of the hearing aid with wireless remote processor, sounds from the ambient environment are picked up by a microphone 12 in the earpiece 10 and transmitted with other information (e.g., data representing the earpiece battery 860 voltage as obtained by an A/D 865 converter and transmitted under control of the earpiece ASIC 845 in the auxiliary bit of many successive replies via the reply data register 825) over a primary two-way wireless link 17 to the RPU 16, where the audio signals are enhanced according to the user's needs by the RPU DSP 948. Signal enhancement is achieved via any of a number of techniques well known in the art that use the capabilities

Claims	Prior Art
	of the general-purpose RPU DSP 948. The enhanced audio signals may be combined with other information (e.g., a synthesized voice, generated within the DSP 948, warning that the aforementioned earpiece battery voltage is low) and transmitted from the RPU 16 over the primary wireless link 17 to the earpiece 10, where they are converted by a speaker 15 to sounds that can only be heard by the user 11. Art auxiliary speaker can be supplied in the RPU 16 as a peripheral 950 device that allows people other than the user to hear the same sounds as those produced by the earpiece speaker 15, if desired (as, e.g., when a person other than the user wishes to use the system's optional cellular telephone capabilities). A different, optional secondary two-way wireless link (shown as part of the RPU in FIG. 9 as secondary wireless link circuitry 944, or connected to the RPU by a wired or other, e.g. infrared, wireless link 18 and shown as optional secondary wireless link circuitry 19 in FIG. 1) can be used for communication between the RPU and a cellular telephone system or other sources of information. An RPU keyboard suitable for covert data entry 72, or voice recognition capabilities in the RPU (implemented, e.g., using any of a number of techniques well known in the art that use the general-purpose RPU DSP 948), can be used to control hearing aid parameters (such as amplification level set by the RPU DSP 948) and telephone dialing functions (e.g., automatic dialing of a predefined telephone number using information stored in the RPU DSP 948) in a manner that can be made imperceptible to a casual observer if desired. Anderson ’783 at 27:4–24 Hearing test: The present invention allows hearing tests to be performed via the earpiece normally worn by the user, without the need for additional expensive test equipment. All signal generation capabilities necessary to perform hearing tests are available in the hearing aid system comprising an earpiece 10 and RPU 16. Specifically, the RPU DSP 948 can be used to generate tones of varying frequencies and amplitudes, as well as other signals known to be useful for hearing test purposes (e.g., nonsense syllables) that are converted to acoustic waves by the earpiece. Such tones are used to perform a hearing test that determines appropriate gain vs. frequency parameters for a program stored in the RPU DSP 948 that performs signal enhancement to compensate for the user's hearing loss. Note that the hearing test program that controls the RPU DSP 948 during the hearing test can be temporarily stored in the RPU DSP 948 for the duration of the test, then deleted upon completion of the test to allow re-use of RPU DSP 948 memory resources during normal operation. The hearing test program may be loaded into the RPU DSP 948 through the secondary wireless link 944 or a wired peripheral link 950. Jannard ’526 at 28:52–64 With reference to FIG. 19, a more detailed illustration of the transceiver 114 is illustrated therein. As shown in FIG. 19, the transceiver includes a preamplifier 230, a band pass filter 232, and an amplifier 234

Claims	Prior Art
	connected in series. The preamplifier 230 and the amplifier 234 can be of any known type, as known to those of ordinary skill in the art. The band filter 232, in the present embodiment, can be constructed as a band pass filter, allowing signals having a frequency from 48 MHz to 50 MHz, inclusive, to pass therethrough. Alternatively, the band filter 232 can be comprised of three band pass filters configured to allow frequencies centered on 48 MHz, 49 MHz, and 50 MHz, respectively, pass therethrough. Jannard '526 at 29:9–19 As the detectors 242, 244 detect 48 MHz and 50 MHz detectors, respectively, they output signals to a spread spectrum pattern detector 250. The spread spectrum pattern detector outputs a corresponding signal to a serial-to-parallel converter 252. The output of the serial-to-parallel converter 252 is output to a digital-to-analog converter 204. A "class D" audio amplifier (not shown), for example, but without limitation, can be connected to the output of the digital-to-audio converter 204 to thereby supply an audio signal to the speakers 14, 14A, 14B, 14C, 14D, 14E, 16, 16A, 16B, 16C, 16D, 16E. Rader '474 at 6:52-60 The effect environmental noise on auditory perception is to reduce the dynamic range available for sounds above the masking level of the environmental noise and to cause abnormal loudness perception of sound heard simultaneously with the noise. Sound ID's Full Frequency Dynamic Compression (FFDC) replicates the compressive function of the healthy cochlea, compressing sound such that the audio signal is no longer distorted in the auditory system and perception is restored to normal. Rader '474 at 6:61-7:18 Compression is functionally an automatic gain control (AGC) within the personal communication device, where the gain reduces as the signal level increases. In order to properly compensate for reduced audibility from the environmental noise, the compression processes the signal in the same way that the cochlea's AGC processes sound in a quiet environment. The input-output function of such a compressor when measured with a quasi-steady-state pure-tone will have a compressive region that has a slope less than or equal to one over a wide dynamic input range (>40 dB), with possible linear regions below and above the compressive region. The gain and compression ratio can vary with frequency and will be monotonic with the noise power in that frequency region. The slope of the compressive region can have values that restore loudness perception to normal in that frequency region, and will have slopes that range from 1 (no compression) to 1/3 (3:1 compression). Because the amount of gain added at any frequency is a function of the power of the signal in that frequency region, the compressor calculates power and applies gain using auditory filter bandwidths that replicate normal critical bands in human hearing. The

Claims	Prior Art
	AGC may be implemented using software executed by a processor in the personal communication device, or by dedicated circuits, or by a combination of these techniques. Rader '474 at 16:30–36 Compression of high dynamic range audio entertainment signals, whether received by the audio system or generated from representative digital or analog storage representative of audio, by systems within the portable entertainment system would provide compensation for the noise induced hearing changes experienced by the listener. Rader '474 at 16:37–44 Mapping of coefficients could take into account base coefficient values that are representative of a hearing impaired individual's requirements for hearing in quiet. Additional compression, whether in frequency ranges where an individual has impairment, or in frequency bands where no impairment compensation in quiet is required, would be beneficial for the additional hearing changes induced by background noise. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Rader to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[60a] A head-worn personal electronic apparatus as recited in claim 59, wherein the head-worn personal electronic apparatus comprises: a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 59, wherein the head-worn personal electronic apparatus comprises: a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user." *See* above at claim [29a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[60b] a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input associated with an environment around the head-worn personal electronic apparatus,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input associated with an environment around the head-worn personal electronic apparatus." *See* above at claim [29b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[60c] wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
61. A head-worn personal electronic apparatus as recited in claim 60, wherein the head-worn personal electronic apparatus comprises: a user input device supported by the head-worn personal electronic apparatus, the user input device being configured to receive a user input, the user input device is based on at least capacitive effect, wherein the user input being received induces at least a capacitive change to produce at least an electrical response.	DHA systems, alone or in view of Bodley and Zadesky or Hollemans, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 60, wherein the head-worn personal electronic apparatus comprises: a user input device supported by the head-worn personal electronic apparatus, the user input device being configured to receive a user input, the user input device is based on at least capacitive effect, wherein the user input being received induces at least a capacitive change to produce at least an electrical response." Bodley '433 at FIG. 1  FIG. 1 Bodley '433 at 2:7–8 FIG. 1 is a frontal view of a generic headset that includes a display in accordance with the invention; Bodley '433 at 2:38–50 Although headset 100 may use any of a variety of means of attaching to a user (e.g., headband, earpiece, etc.) and thus allowing hands-free operation of the headset, preferably headset 100 includes an earpiece 105 (e.g., an ear hook) that is coupled to the user's ear, or a portion thereof. More preferably, earpiece 105 fits around a portion of the user's ear. Earpiece 105 is coupled to body 107 of the headset, body 107 housing the various electronic components necessary to provide functionality to headset 100. Attached to the back surface of body 107 is an output transducer 109 (e.g., a speaker). Also coupled to body 107 is an

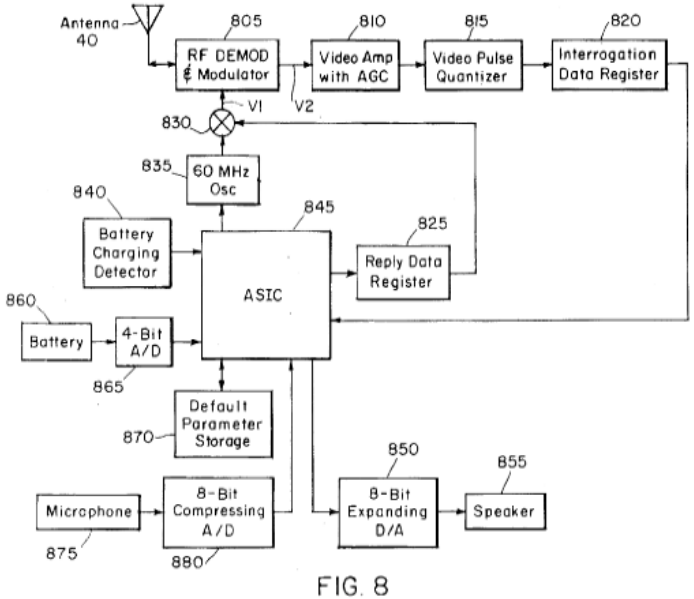
Claims	Prior Art
	input transducer 111 (e.g., a microphone), which, in this embodiment, is located at the end of a boom member 113. Bodley ’433 at 2:65–3:22 Display 103 can be used to provide the user with various types of information depending upon the design goals for the specific headset. The design goals are primarily based on the expected headset use, desired headset functionality, display size, display resolution, allowable headset power drain, and the processing capabilities of the headset on-board processor (also referred to as a controller or an interface controller). Examples of information that can be displayed on display 103 are headset battery level, headset signal level, call status (e.g., dialing, in use, etc.), caller identification, current time/date, and timer information such as elapsed time associated with a particular call. Display 103 can also be configured to flash or otherwise visually indicate an incoming call. Assuming that the information is provided to the headset processor, display 103 can also display the battery level of the base unit (e.g., cell phone, dongle, etc.) and the signal strength of the base unit (e.g., cell phone). Additionally, display 103 can be used during headset configuration (e.g., to adjust or set the ring mode, ring loudness, headset volume, display mode, voice dialing, time, date, etc.) or to provide additional headset functionality (e.g., phone lists, text messages, dialing using virtual numeric keypad or phone list dialing). Additionally, display 103 can be touch sensitive, thus providing a touch screen for data input (e.g., phone numbers into a phone list, dialing using a virtual numeric keypad). Zadesky ’230 at 2:16–59 With regards to touch pads, mice and track balls, a Cartesian coordinate system is used to monitor the position of the finger, mouse and ball, respectively, as they are moved. The Cartesian coordinate system is generally defined as a two dimensional coordinate system (x, y) in which the coordinates of a point (e.g., position of finger, mouse or ball) are its distances from two intersecting, often perpendicular straight lines, the distance from each being measured along a straight line parallel to each other. For example, the x, y positions of the mouse, ball and finger may be monitored. The x, y positions are then used to correspondingly locate and move the input pointer on the display screen. To elaborate further, touch pads generally include one or more sensors for detecting the proximity of the finger thereto. By way of example, the sensors may be based on resistive sensing, surface acoustic wave sensing, pressure sensing (e.g., strain gauge), optical sensing, capacitive sensing and the like. The sensors are generally dispersed about the touch pad with each sensor representing an x, y position. In most cases, the sensors are arranged in a grid of columns and rows. Distinct x and y position signals, which control the x, y movement of a pointer device on the display screen, are thus generated when a finger is moved

Claims	Prior Art
	across the grid of sensors within the touch pad. For brevity sake, the remaining discussion will be held to the discussion of capacitive sensing technologies. It should be noted, however, that the other technologies have similar features. Capacitive sensing touch pads generally contain several layers of material. For example, the touch pad may include a protective shield, one or more electrode layers and a circuit board. The protective shield typically covers the electrode layer(s), and the electrode layer(s) is generally disposed on a front side of the circuit board. As is generally well known, the protective shield is the part of the touch pad that is touched by the user to implement cursor movements on a display screen. The electrode layer(s), on the other hand, is used to interpret the x, y position of the user's finger when the user's finger is resting or moving on the protective shield. The electrode layer (s) typically consists of a plurality of electrodes that are positioned in columns and rows so as to form a grid array. The columns and rows are generally based on the Cartesian coordinate system and thus the rows and columns correspond to the x and y directions. Zadesky ’230 at 2:60–3:5 The touch pad may also include sensing electronics for detecting signals associated with the electrodes. For example, the sensing electronics may be adapted to detect the change in capacitance at each of the electrodes as the finger passes over the grid. The sensing electronics are generally located on the backside of the circuit board. By way of example, the sensing electronics may include an application specific integrated circuit (ASIC) that is configured to measure the amount of capacitance in each of the electrodes and to compute the position of finger movement based on the capacitance in each of the electrodes. The ASIC may also be configured to report this information to the computing device. Zadesky ’230 at 3:6–22 Referring to FIG. 1, a touch pad 10 will be described in greater detail. The touch pad is generally a small rectangular area that includes a protective shield 12 and a plurality of electrodes 14 disposed underneath the protective shield layer 12. For case of discussion, a portion of the protective shield layer 12 has been removed to show the electrodes 14. Each of the electrodes 14 represents a different x, y position. In one configuration, as a finger 16 approaches the electrode grid 14, a tiny capacitance forms between the finger 16 and the electrodes 14 proximate the finger 16. The circuit board/sensing electronics measures capacitance and produces an x, y input signal 18 corresponding to the active electrodes 14 is sent to a host device 20 having a display screen 22. The x, y input signal 18 is used to control the movement of a cursor 24 on a display screen 22. As shown, the input pointer moves in a similar x, y direction as the detected x, y finger motion.

Claims	Prior Art
	Zadesky ’230 at 12:31–41 The capacitance in each of the electrodes 258 is measured by control circuitry 260 located on the backside of the circuit board 256. By detecting changes in capacitance at each of the electrodes 258, the control circuitry 260 can determine the angular location, direction, speed and acceleration of the finger as it is moved across the touch pad 250. The control circuitry 260 can also report this information in a form that can be used by a computing device. By way of example, the control circuitry may include an ASIC (application specific integrated circuit). Hollemans ’490 at Abstract In a personal audio system (100), a remote controller (120) has a touch-sensitive area (122) and is worn in or by a human ear. A temporal pattern in the area (122), which is being touched, is detected and used for remotely controlling a device (110) for personal audio by means of a control signal (130). This prevents the hassle involved in finding, manipulating and operating a conventional remote control that is typically dangling somewhere along a wire. Hollemans ’490 at 4:30–5:7 The outer surface 121 is just the physical outer side of the controller 120. A part of this outer surface 121 is sensitive to being touched and is referred to as the touch-sensitive area 122. Optionally, the controller 120 comprises a second area that is sensitive to being touched, referred to as the further touch-sensitive area 123. The touch-sensitive area 122 can be realized in many ways, including a sensor for changes in capacity or induced voltages or currents, a grid of conducting parts between which changes in resistance are measured and detected, or an ordinary pressure sensor, button or temperature sensor. Another possibility is to measure and detect changes in either a passive or an active electromagnetic field. With the passive variant, the controller 120 relies on, for example, nearby power lines to generate measurable fields when being touched. With the active variant, the controller 120 generates an electromagnetic field of its own. Hollemans ’490 at 8:19–22 In a third way, the touch-sensitive area is used for picking up a gesture made on the area with e.g. a stylus or a finger. The remote control may be extended for this purpose with means for detecting a gesture, e.g. a laser beetle, a small camera, an IR movement sensor, or a cross-capacitance sensor.

Claims	Prior Art
	Hollemans ’490 at 8:30–33 The gesture detected may be a user defined gesture for activating a user defined function. The gesture detected may be used for user identification by matching with initials or an autograph. The gesture may extend into two dimensions, e.g. a planar gesture, or it may extend into three dimensions. A POSITA would have been motivated to combine the teachings of the DHA system and Bodley, or Zadesky or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 3, 5, and 6. To the extent that DHA systems, alone or in view or Bodley, Zadesky, or Hollemans do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
62. A head-worn personal electronic apparatus as recited in claim 61, wherein the electronic circuitry is electrically connected to the user input device to receive user input data from the user input device, and wherein the head-worn personal electronic apparatus is configured to distinguish at least whether the user input pertains to a forward sliding action or a backward sliding action with respect to the user input device using at least the electronic circuitry and the user input data.	DHA systems, in view of Zadesky or Hollemans, expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 61, wherein the electronic circuitry is electrically connected to the user input device to receive user input data from the user input device, and wherein the head-worn personal electronic apparatus is configured to distinguish at least whether the user input pertains to a forward sliding action or a backward sliding action with respect to the user input device using at least the electronic circuitry and the user input data.” *See* above at claim 24. A POSITA would have been motivated to combine the teachings of the DHA system and Zadesky or Hollemans to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 3, 6, and 24. To the extent that DHA systems, alone or in view of Zadesky or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
63. A head-worn personal electronic apparatus as recited in claim 61, wherein the head-worn wearable electronic device includes an accelerometer.	DHA systems, alone or in view of Jannard or Abreu or Irvin, expressly or inherently disclose “[a] head-worn personal electronic apparatus as recited in claim 61, wherein the head-worn wearable electronic device includes an accelerometer.” Jannard ’526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is

Claims	Prior Art
	provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance. Abreu ’960 at 13:46–60 The eyeglasses or other support structures can also have elements which produce and radiate recognizable signals and this procedure could be used to locate and track individuals, particularly in military operations. A permanent magnet can also be mounted in the eyeglasses and used for tracking as described above. A fixed frequency transmitter can be mounted in the eyeglasses and used as a tracking device which utilizes a satellite tracking system by noting the frequency received from the fixed frequency transmitter to a passing satellite, or via Global Positioning Systems. Motion and deceleration can be detected by mounting an accelerometer in the eyeglasses. The use of eyeglasses as tracking means can be useful for locating a kidnapped individual or for rescue operations in the military, since eyeglasses are normally unsuspecting articles. Irvin ’332 at 5:1–23 Because the wireless headset 100 of the present invention automatically enables and disables its communication functions based on use, the sensor 220 may be based on a variety of technologies. In general, its purpose is to provide a signal responsive to a natural stimulus associated with use of the headset by a user. Movement or vibration of the wireless headset 100, or placement of the wireless headset 100 upon the user's body indicates usage. Movement may be detected based on vibration or by sensing acceleration, such as with an accelerometer. Mechanical or liquid-based switches represent alternate mechanisms for sensing movement. In such switches, movement causes an electrical connection to be established between contact pairs. Further, the particular sensor 220 employed may be tuned or configured such that only stimulus consistent with actual use actuates the sensor, thus avoiding inadvertent or unintended enabling of the wireless headset 100. In such configurations, the sensor 220 preferably employs output conditioning that prevents assertion of the sensor output signal for stimulus below a defined threshold. Alternatively, the power control circuit 210 may include such conditioning as part of its interface to the sensor 220. FIGS. 4A–4D illustrate possible sensor embodiments. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Jannard or Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
64. A head-worn personal electronic apparatus as recited in claim 63, wherein the head-worn personal electronic apparatus comprises: a radiation sensor, operatively connected to the rechargeable battery, at least partially contained within the head-worn personal electronic apparatus and configured to sense radiation.	DHA systems, alone or in view of Anderson, Jannard, or Abreu, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 63, wherein the head-worn personal electronic apparatus comprises: a radiation sensor, operatively connected to the rechargeable battery, at least partially contained within the head-worn personal electronic apparatus and configured to sense radiation." Phonak MicroLink MLxS Product Information at Page. 1 • Powered by the hearing instrument battery Anderson '783 at FIG. 8  FIG. 8 Anderson '783 at 3:40–41 FIG. 8 is a block diagram showing details of an earpiece for a full-featured preferred embodiment of the invention.

Claims	Prior Art
	Anderson ’783 at 6:55–57 Earpiece power consumption is preferably minimized to extend battery life, and use of a rechargeable battery is desirable. Anderson ’783 at 24:15–26 The invention provides useful unsolicited information (i.e., information automatically provided to the user by the system, not as a result of a user request for information) to the user of the system in a manner that can be made imperceptible to a casual observer if desired. Information about the status of the hearing aid system itself, such as a verbal warning that an earpiece or RPU battery voltage is low, is provided to the user as detailed earlier. Automatic measurements of heart pulse rate, body temperature and walking distance traveled can be taken by sensors 950 in the RPU 16 or earpiece 10 and stored in the RPU DSP 948. Jannard ’526 at 20:57–64 The interface device 110 includes a power source 112, a transceiver 114, an interface 116, and an antenna 118. The power source 112 can be in the form of disposable or rechargeable batteries. Optionally, the power source 112 can be in the form of solar panels and a power regulator. The transceiver 114 can be in the form of a digital wireless transceiver for one-way or two-way communication. Jannard ’526 at 24:37–55 Optionally, the power source 112 can comprise portions of the ear stems 54D, 56D which have been formed into batteries. For example, the rear portions 160, 162 of the ear stems 54D, 56D, respectively, can be in the form of custom made batteries, either disposable or rechargeable. Preferably, the rear portions 160, 162 are removable from the forward portions of the ear stems 54D, 56D. This provides a particular advantage in terms of balance. As noted above, imbalanced loads on the head can cause muscular pain and/or headaches. In particular, excessive pressure on the nose can cause severe headaches. Additionally, batteries can have a significantly higher mass density than plastic and lightweight metals, such as sintered titanium or magnesium. Thus, by constructing the rearward portions 160, 162 of the ear stems 54D, 56D of batteries, the weight of these batteries can improve forward-rearward balance of the audio device 10D in that the weight of the interface device 110 can be offset by the batteries. In another embodiment, the ear stems 54D, 56D can define a housing for removable batteries.

Claims	Prior Art
	Jannard ’526 at 24:56–25:3 The audio device 10D can also include power contacts 164 for recharging any rechargeable batteries connected thereto. For example, the power contacts 164 can be disposed on a lower edge of the orbitals 48D, 50D. Thus, with an appropriate recharging cradle (not shown), the audio device 10D can be laid on the cradle, thereby making contact between the power contacts 164 and corresponding contacts in the cradle (not shown). Alternatively, power contacts can be provided in numerous other locations as desired. For example, the power contacts 164 can be disposed at the ends of the ear stems 54D, 56D. A corresponding cradle can include two vertically oriented holes into which the ear stems are inserted for recharging. In this configuration, the lens within the orbitals 48D, 50D would face directly upwardly. Jannard ’526 at 26:21–27 With reference to FIGS. 8, 9A, and 9B, in another embodiment, the transceiver 114 is adapted to employ a wide variety of technologies, including wireless communication such as RF, IR, ultrasonic, laser or optical, as well as wired and other communications technologies. In one embodiment, a body-LAN radio is employed. Other embodiments can employ a flexible-circuit design. Jannard ’526 at 26:46–51 In another embodiment, a further modification of the audio devices 10, 10A, 10B, 10C, and 10D is provided wherein the audio devices can include and/or communicate with a variety of sensors, including but not limited to motion, radar, heat, light, smoke, air-quality, oxygen, CO and distance. Abreu ’960 at 6:28–38 Head-mounted structures are structures mounted on the head or neck for positioning sensors on the end of the tunnel and include head bands with accessories that are adjacent to the tunnel, visors, helmets, headphone, structures wrapping around the ear and the like. For the purpose of this description TempAlert is referred herein as a system that measures temperature in the BTT area and has means to report the measured value and that can incorporate alarm means that are activated when certain levels are reached. Support structures yet include any article that has sensing means in which said sensing means are positioned at the end of the tunnel. Abreu ’960 at 10:49–55 The apparatus of the invention include a temperature sensor located in the medial canthal pad. For temperature measurement the sensing system is located on a skin area that includes the medial canthal

Claims	Prior Art
	corner of the eye and upper eyelid. The sensor in the medial canthal pad is preferably positioned on the skin adjacent to the medial canthal area (medial corner of the eye). Abreu '960 at 32:29–45 In accordance with this exemplary embodiment for temperature measurement, the thermal energy emitted by the BTT is sensed by the temperature sensor 102 such as a miniature thermistor which produces a signal representing the thermal energy sensed. The signal is then converted to digital information and processed by processor 106 using standard processing for determining the temperature. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Abreu to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Abreu, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65pre] A head-worn personal electronic apparatus operable to at least output customized audio output, comprising:	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, and Bodley '433 or Hollemans '490, and Nemirovski '783 and Abreu '960 or Irvin '332, expressly or inherently disclose "[a] head-worn personal electronic apparatus operable to at least output customized audio output." *See* above at claim 1[pre], [17pre], [3], [7a]. [7b], [13]. To the extent that DHA systems, alone or in view of the above cited reference(s), do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65a] at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user;	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user." *See* above at claim [1a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65b] a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear;	DHA systems, alone or in view of Anderson Enzmann, Jannard, or Rader, expressly or inherently disclose "a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear." *See* above at claim [17b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, or Rader to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65c] electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker;	DHA systems, alone or in view of Anderson, Enzmann, Rader, or Jannard expressly or inherently disclose "electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker." *See* above at claim [17c]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, or Enzmann, Jannard, or Rader to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65d] a touch sensitive surface at least partially contained within the	DHA systems, alone or in view of Bodley or Hollemans, expressly or inherently disclose "a touch sensitive surface at least partially contained within the head-worn personal electronic apparatus and configured to sense

Claims	Prior Art
head-worn personal electronic apparatus and configured to sense a touch input associated with the user, wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus;	a touch input associated with the user, wherein the sensed touch input operating to at least alter operation of the head-worn electronic personal audio apparatus." *See* above at claim [3a] and [3b]. A POSITA would have been motivated to combine the teachings of the DHA system and Bodley to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Bodley, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65e] a sensor at least partially contained within the head-worn personal electronic structure and configured to sense data indicative of whether the head-worn personal electronic apparatus is being worn by the user, the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the sensed data from the sensor;	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "a sensor at least partially contained within the head-worn personal electronic structure and configured to sense data indicative of whether the head-worn personal electronic apparatus is being worn by the user, the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the sensed data from the sensor." *See* above at claims [13] and [14]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 13. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65f] a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user." *See* above at claim [7a].

Claims	Prior Art
personal electronic apparatus is being worn by the user; and	A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65g] a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[65h] wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals.	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals." *See* above at claims [1f], [17d]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[66a] A head-worn personal electronic apparatus as recited in claim 65, wherein the sensor comprises a radiation detector configured to capture radiation data, and	DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the sensor comprises a radiation detector configured to capture radiation data." *See* above at claim [13a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[66b] wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data.	DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, expressly or inherently disclose "wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data." *See* above at claims [13b] and [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Abreu or Luo or Irvin to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Anderson or Abreu or Luo or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
67. A head-worn personal electronic apparatus as recited in claim 65, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested	DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation." *See* above at claim [54b].

Claims	Prior Art
operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation.	To the extent that DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
68. A head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone.	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone." *See* above at claim [11e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
69. A head-worn personal electronic apparatus as recited in claim 65, wherein the ambient sound picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound picked-up from the second microphone is subtracted from the audio picked-up from the first microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the ambient sound picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound picked-up from the second microphone is subtracted from the audio picked-up from the first microphone." *See* above at claim [11e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[70a] A head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. 7, and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[70b] modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level.	DHA systems, alone or in view of Nemirovski and Victorian, expressly or inherently disclose "modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level." *See* above at claim [38d]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski and Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
72. A head-worn personal electronic apparatus recited in claim 65, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user.	DHA systems, alone or in view of Jannard, Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus recited in claim 65, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user." *See* above at claim 27. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Abreu or Irvin, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 3, 13, and 27.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Jannard or Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[75a] A head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 65, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input." *See* above at claim [9a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[75b] provide at least the ambient audio output signal to the at least one speaker.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "provide at least the ambient audio output signal to the at least one speaker." *See* above at claim [9b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[76a] A head-worn personal electronic apparatus as recited in claim 75, wherein the electronic circuitry is configured to: determine an ambient noise level	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 75, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a].

Claims	Prior Art
based on at least the ambient sound input,	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[76b] wherein the ambient audio output signal is dependent on the ambient noise level.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the ambient audio output signal is dependent on the ambient noise level." *See* above at claim [10b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77pre]. A head-worn personal electronic apparatus operable to at least output customized audio output, comprising:	DHA systems, alone or in view of Anderson, Enzmann, Rader, or Jannard, Nemirovski '783 and Abreu '960 or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus operable to at least output customized audio output." *See* above at claim 1[pre], [17pre], [7a]. [7b], [14]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[77a] at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user;	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose "at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user." *See* above at claim [1a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77b] a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear;	DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, expressly or inherently disclose "a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear." *See* above at claim [17b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, or Rader, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77c] electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker;	DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, expressly or inherently disclose "electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker." *See* above at claim [17c]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, or Rader, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1.

Claims	Prior Art
	To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77d] a force sensor at least partially contained within the head-worn personal electronic apparatus and configured to receive a user input based at least on sensing force applied to the force sensor, the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device;	DHA systems, alone or in view of Abreu or Irvin, expressly or inherently disclose "a force sensor at least partially contained within the head-worn personal electronic apparatus and configured to receive a user input based at least on sensing force applied to the force sensor, the force sensor is configured to detect the user input, the detected user input operating to alter at least an operation of the head-worn wearable electronic device." *See* above at claim [14a] and [14b]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 13, and 14. To the extent that DHA systems, alone or in view of Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77e] a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user; and	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a first microphone configured to pick up at least audio input from at least the user, the first microphone being at least partially in the head-worn personal electronic apparatus and is positioned to be directed towards the user when the head-worn personal electronic apparatus is being worn by the user." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[77f] a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone configured to pick up at least ambient sound input associated with an environment in the vicinity of the head-worn personal electronic apparatus, the second microphone being positioned to be directed away from the user when the head-worn wearable electronic device is being worn by the user." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[77g] wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals.	DHA systems, alone or in combination with Anderson, Enzmann, Rader, or Jannard, expressly or inherently disclose "wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals." *See* above at claims [1f], [17d]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Rader or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in combination with Anderson, Enzmann, Rader, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[78a] A head-worn personal electronic apparatus as recited in claim 77, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data,	DHA systems, alone or in view of Abreu or Irvin or Luo, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the head-worn personal electronic apparatus comprises: a radiation detector configured to capture radiation data." *See* above at claim [13a]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin or Luo, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13.

Claims	Prior Art
	To the extent that DHA systems do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[78b] wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data.	DHA systems, alone or in view of Abreu or Irvin or Luo, expressly or inherently disclose "wherein the electronic circuitry is configured to process at least the radiation data captured by the radiation detector, and configured to determine whether the head-worn personal electronic apparatus is being worn by a user based on at least the processed radiation data." *See* above at claim [13b] and [13c]. A POSITA would have been motivated to combine the teachings of the DHA system and Abreu or Irvin or Luo, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 13. To the extent that DHA systems, alone or in view of Abreu or Irvin or Luo, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
79. A head-worn personal electronic apparatus as recited in claim 77, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation.	DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation." *See* above at claim [54b]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard or Nemirovski or Seshadri, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson or Jannard or Nemirovski or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
80. A head-worn personal electronic apparatus as recited in	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the electronic circuitry

Claims	Prior Art
claim 77, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone.	is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
81. A head-worn personal electronic apparatus as recited in claim 77, wherein the ambient sound input picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound input picked-up from the second microphone is subtracted from the audio picked-up from the first microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the ambient sound input picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound input picked-up from the second microphone is subtracted from the audio picked-up from the first microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[82a] A head-worn personal electronic apparatus as recited in claim 77, wherein the electronic circuitry is configured to: determine an ambient noise level	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a].

Claims	Prior Art
based on at least the ambient sound input; and	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[82b] modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level.	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level." *See* above at claim [8b] and [38d]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 7, and 8. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[85a] A head-worn personal electronic apparatus as recited in claim 77, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 77, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input." *See* above at claim [9a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[85b] provide at least the ambient audio output signal to the at least one speaker.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "provide at least the ambient audio output signal to the at least one speaker." *See* above at claim [9b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[86a] A head-worn personal electronic apparatus as recited in claim 85, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input,	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 85, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[86b] wherein the ambient audio output signal is dependent on the ambient noise level.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the ambient audio output signal is dependent on the ambient noise level." *See* above at claim [10b]. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[87pre] A head-worn personal electronic apparatus operable to at least output customized audio output, comprising:	DHA systems, alone or in view of Anderson, Enzmann, Rader, Jannard, Bodley ’433 or Hollemans ’490, and Nemirovski ’783 and Abreu ’960 or Irvin ’332, expressly or inherently disclose “[a] head-worn personal electronic apparatus operable to at least output customized audio output.” *See* above at claim 1[pre]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, Bodley ’433, Hollemans ’490, Nemirovski ’783, Abreu ’960 and/or or Irvin ’332 to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1, 17, 65, 77, and 87. To the extent that DHA systems, alone or in view of the above cited references, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87a] at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user;	DHA systems, alone or in view of Anderson, Enzmann, or Jannard, expressly or inherently disclose “at least one speaker, the at least one speaker being configured to be wearable proximate to an ear of a user.” *See* above at claim [1a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87b] a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized	DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, expressly or inherently disclose “a wireless receiver configured to wirelessly receive customized audio output signals for the user, the customized audio output signals having been customized based on at least predetermined hearing characteristics associated with the user such that the customized audio output signals at least compensate for the hearing impairment regarding the user's ability to hear.” *See* above at claim [17b].

Claims	Prior Art
audio output signals at least compensate for the hearing impairment regarding the user's ability to hear;	A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, or Rader, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87c] electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker;	DHA systems, alone or in view of Anderson, Ezmann, Jannard, or Rader, expressly or inherently disclose "electronic circuitry including at least an integrated circuit, the electronic circuitry being configured to output the customized audio output signals to the at least one speaker." *See* above at claim [17c]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, Jannard, or Rader, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, Jannard, or Rader, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87d] a user input device at least partially contained within the head-worn personal electronic apparatus and configured to receive a user input, the user input operating to alter at least an operation of the head-worn wearable electronic device;	DHA systems, alone or in view of Bodley, Zadesky, or Hollemans, expressly or inherently disclose "a user input device at least partially contained within the head-worn personal electronic apparatus and configured to receive a user input, the user input operating to alter at least an operation of the head-worn wearable electronic device." *See* above at claims [3a] and [3b]. A POSITA would have been motivated to combine the teachings of the DHA system and Bodley, Zadesky, or Hollemans, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 3. To the extent that DHA systems, alone or in view of Bodley, Zadesky, or Hollemans, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[87e] a radiation detector at least partially contained within the head-worn personal electronic structure and configured to configured to capture radiation data indicative of whether the head-worn personal electronic apparatus is being worn by the user, the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the captured radiation data;	DHA systems, alone or in view of Jannard, or Abreu or Irvin, expressly or inherently disclose "a radiation detector at least partially contained within the head-worn personal electronic structure and configured to configured to capture radiation data indicative of whether the head-worn personal electronic apparatus is being worn by the user, the head-worn personal electronic apparatus is configured to at least alter operation of the head-worn personal electronic apparatus based at least in part on the captured radiation data." *See* above at claim [65e]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard, or Abreu or Irvin, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 13. To the extent that DHA systems, alone or in view of Jannard, or Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87f] a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user; and	DHA systems, alone or in view of Andrson, Jannard, or Nemirovski, expressly or inherently disclose "a first microphone at least partially contained within the head-worn personal electronic apparatus, the first microphone being positioned to pick up at least audio input from at least the user." *See* above at claim [7a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87g] a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, expressly or inherently disclose "a second microphone at least partially contained within the head-worn personal electronic apparatus, the second microphone being positioned to pick up at least environmental audio input associated with an environment around the head-worn personal electronic apparatus." *See* above at claim [7b].

Claims	Prior Art
associated with an environment around the head-worn personal electronic apparatus,	A POSITA would have been motivated to combine the teachings of the DHA system and Anderson ,Jannard, or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[87h] wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals.	DHA systems, alone or in view of Anderson, Enzmann, or Jannard expressly or inherently disclose "wherein the at least one speaker is configured to generate at least a portion of the customized audio output for the user in accordance with at least the portion of the customized audio output signals." *See* above at claim [1f], [17d]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Enzmann, or Jannard to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson, Enzmann, or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
88. A head-worn personal electronic apparatus as recited in claim 87, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Seshadri, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the head-worn personal electronic apparatus wherein the head-worn personal electronic device is configured to receive a voice input via the first microphone, determine a requested operation from the voice input using voice recognition, and control an operation by the head-worn personal electronic device based on the requested operation." *See* above at claim [54a]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Seshadri, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claim 1. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
89. A head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the ambient sound input picked-up via the second microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
90. A head-worn personal electronic apparatus as recited in claim 87, wherein the ambient sound input picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound input picked-up from the second microphone is subtracted from the audio picked-up from the first microphone.	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the ambient sound input picked-up from at least the second microphone is used at least for noise cancelation, and at least a portion of the ambient sound input picked-up from the second microphone is subtracted from the audio picked-up from the first microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[91a] A head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to:	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input."

Claims	Prior Art
determine an ambient noise level based on at least the ambient sound input; and	*See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[91b] modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level.	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "modify at least a portion of the audio sound picked-up via the first microphone, with the modification being at least in part dependent on the ambient noise level." *See* above at claim [38d]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[92a] A head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to: determine an environmental sound level based on at least the environmental audio input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to: determine an environmental sound level based on at least the environmental audio input." *See* above at claim [38c]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[92b] amplify at least a portion of the audio input from the first microphone, with the amplification being at least in part dependent on the environmental sound level.	DHA systems, alone or in view of Nemirovski or Victorian, expressly or inherently disclose "amplify at least a portion of the audio input from the first microphone, with the amplification being at least in part dependent on the environmental sound level." *See* above at claim [38d]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[95a] A head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input; and	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to: produce an ambient audio output signal based on at least the ambient sound input." *See* above at claim [9a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[95b] provide at least the ambient audio output signal to the at least one speaker.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "provide at least the ambient audio output signal to the at least one speaker." *See* above at claim [9b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with

Claims	Prior Art
	the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[96a] A head-worn personal electronic apparatus as recited in claim 95, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input,	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 95, wherein the electronic circuitry is configured to: determine an ambient noise level based on at least the ambient sound input." *See* above at claim [8a]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[96b] wherein the ambient audio output signal is dependent on the ambient noise level.	DHA systems, alone or in view of Jannard or Victorian, expressly or inherently disclose "wherein the ambient audio output signal is dependent on the ambient noise level." *See* above at claim [10b]. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 8. To the extent that DHA systems, alone or in view of Jannard or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
97[pre]. A head-worn personal electronic apparatus as recited in claim 87, wherein head-worn personal electronic apparatus is configured to support voice calls to the user, and	DHA systems, alone or in view of Jannard or Bodley or Hollemans or Seshadri, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein head-worn personal electronic apparatus is configured to support voice calls to the user." Jannard '526 at 24:29–36 In a further illustrative and non-limiting example, the medial-lateral movement of the button 152, along the directions identified by the arrow 160, can be used to choose different functions performed by the transceiver 114 or the interface 116. For example, an inward movement of the button 152 could be used to

Claims	Prior Art
	answer an incoming phone call where the audio device 10D is used as an audio interface for a cellular phone. Jannard ’526 at 30:40–52 The headset profile is designed to be used for interfacing a headset having one earphone, a microphone, and a transceiver worn by the wearer, for example, on a belt clip, with a cordless phone through a wireless connection. According to the headset profile, certain commands can be issued from a headset, such as the audio devices 10, 10A, 10A′, 10B, 10C, 10D, and 10E, using an AT command protocol. In such a protocol, text commands must be input to the BLUETOOTH™ device, which the BLUETOOTH™ device then transmits wirelessly to a synchronized BLUETOOTH™ enabled device. Such commands include, for example, but without limitation, initiating a call, terminating a call, and redialing a previously dialed number. Bodley ’433 at 1:47–52 The present invention provides a wireless headset with an integral display, the headset capable of communicating via a wireless network to a cellular telephone, cellular telephone adaptor, land-line telephone, land-line telephone adaptor, computer, personal digital assistant, or other device capable of communicating via the wireless network. Bodley ’433 at 1:52–57 The wireless headset of the invention includes an input transducer (e.g., a microphone), an output transducer (e.g., a speaker), a wireless networking subsystem and a controller/controller interface. Preferably the headset also includes means for attaching the headset to the user in order to allow hands-free operation. Hollemans ’490 at 9:4–17 Another way of switching between applications may not be initiated by the user, but by the system, e.g., on an incoming phone call or some other event. As soon as the incoming phone call is detected, the ring tone may be sounded, potentially mixed with the active sound source e.g. MP3 playback. While the incoming phone call is communicated to the user, the controls are mapped to the phone application rather than the audio playback application. In the phone application, touching the right earpiece could be mapped on answering the call and the left earpiece could be mapped to hanging up. As the user picks up the call, the playback of the music may be paused.

Claims	Prior Art
	Seshadri ’839 at [0042] Input module 128, which may be a keypad, touch screen, voice recognition circuit, or other like user interfaces, receives user commands and produces digital command messages 124 there from. Such digital command messages 124 includes, but are not limited to, packet size, synchronization information, frequency hopping initiation information, timeslot allocation information, link establishment information, piconet address information, fast-forward, play, pause, volume adjust, record, stop and rewind. Seshadri ’839 at [0047] Detachable ear-piece 156 and microphone 154 may have on-chip operations to support call conferencing, call waiting, flash, and other features associated with telephones. These functions may me accessed and reviewed by a user interface 158 and display 160 within host device 152 or a user interface and display located on either detachable ear-piece 156 or microphone 154. The user interface and display, located on either the host device or detachable ear-piece 156 and microphone 154 may have a display and button(s) that may be used to program device, perform directory functions including selecting number to call, view caller ID, initiate call waiting, or initiate call conferencing. Additionally, circuitry within the ear-piece 156 and microphone 154 may enable voice activated dialing. The actual voice recognition could be performed within ear-piece 156, microphone 154, or host device 152. Thus, ear-piece 156 and microphone 154 may act to initiate calls and receive calls. A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Bodley or Hollemans or Seshadri, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Jannard or Bodley or Hollemans or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[97a] wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns at least accepting or declining a voice call to the user.	DHA systems, alone or in view of Jannard or Bodley or Hollemans or Seshadri, expressly or inherently disclose “wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns at least accepting or declining a voice call to the user.” *See* above at claim 97[pre].

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Bodley or Hollemans or Seshadri, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Jannard or Bodley or Hollemans or Seshadri, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
98. A head-worn personal electronic apparatus recited in claim 87, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user.	DHA systems, alone or in view of Bodley or Abreu or Irvin, expressly or inherently disclose "[a] head-worn personal electronic apparatus recited in claim 87, wherein the at least one operation of the head-worn personal electronic apparatus being altered concerns control of the customized audio output being provided to the user." *See* above at claims 5 and 17. A POSITA would have been motivated to combine the teachings of the DHA system and Bodley or Abreu or Irvin, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 5 and 27. To the extent that DHA systems, alone or in view of Bodley or Abreu or Irvin, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[99a] A head-worn personal electronic apparatus as recited in claim 87, wherein the head-worn personal electronic apparatus is configured to provide audio alerts to the user via the at least one speaker, and	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the head-worn personal electronic apparatus is configured to provide audio alerts to the user via the at least one speaker." *See* above at claim 51. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.

Claims	Prior Art
[99b] wherein the audio alerts are dependent on one or more user preference settings.	DHA systems, alone or in view of Anderson or Jannard, expressly or inherently disclose "wherein the audio alerts are dependent on one or more user preference settings." *See* above at claim 52. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson or Jannard, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1. To the extent that DHA systems, alone or in view of Anderson or Jannard, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[100a] A head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone,	DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, expressly or inherently disclose "[a] head-worn personal electronic apparatus as recited in claim 87, wherein the electronic circuitry is configured to modify the customized audio output signals to be output to the user via the at least one speaker to cancel at least a portion of noise associated with the environment, the modification of the customized audio output signals being dependent at least in part on the environmental audio input picked-up via the second microphone." *See* above at claims [11e] and [34e]. A POSITA would have been motivated to combine the teachings of the DHA system and Anderson, Jannard, or Nemirovski or Victorian, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Anderson, Jannard, or Nemirovski or Victorian, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[100b] wherein, when the head-worn personal electronic apparatus is being worn by the user, the first microphone is positioned to be directed more towards the user's mouth than is the second microphone, and	DHA systems, alone or in view of Jannard or Nemirovski, expressly or inherently disclose "wherein, when the head-worn personal electronic apparatus is being worn by the user, the first microphone is positioned to be directed more towards the user's mouth than is the second microphone." *See* above at claim [7a].

Claims	Prior Art
	A POSITA would have been motivated to combine the teachings of the DHA system and Jannard or Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Jannard or Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.
[100c] wherein, when the head-worn personal electronic apparatus is being worn by the user, the second microphone is directed more away from the user's mouth than is the first microphone.	DHA systems, alone or in view of Nemirovski, expressly or inherently disclose "wherein, when the head-worn personal electronic apparatus is being worn by the user, the second microphone is directed more away from the user's mouth than is the first microphone." *See* above at claim [7b]. A POSITA would have been motivated to combine the teachings of the DHA system and Nemirovski, to meet the claim limitations with reasonable expectation of success for the reasons described above with respect to claims 1 and 7. To the extent that DHA systems, alone or in view of Nemirovski, do not disclose any elements of this claim limitation, it would have been obvious to a person having ordinary skill in the art to combine, with the teachings of DHA systems, those elements, as known to them or as disclosed by other prior art identified in the cover pleading.