ROHDE & SCHWARZ

Test and Measurement
Division

Supplement to

AUDIO ANALYZER

R&S UPL16

DC to 110 kHz

1078.2008.16

Software version UPL 3.01

Testcases with CRTx: U8-Software 2.00

Testcases with CMU: U81-Software 1.00

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Dear Customer,

The Audio Analyzer R&S UPL16 is abbreviated as UPL16.

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Before putting the product into operation for the first time, make sure to read the following

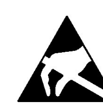
Safety Instructions

Rohde & Schwarz makes every effort to keep the safety standard of its products up to date and to offer its customers the highest possible degree of safety. Our products and the auxiliary equipment required for them are designed and tested in accordance with the relevant safety standards. Compliance with these standards is continuously monitored by our quality assurance system. This product has been designed and tested in accordance with the EC Certificate of Conformity and has left the manufacturer's plant in a condition fully complying with safety standards. To maintain this condition and to ensure safe operation, observe all instructions and warnings provided in this manual. If you have any questions regarding these safety instructions, Rohde & Schwarz will be happy to answer them.

Furthermore, it is your responsibility to use the product in an appropriate manner. This product is designed for use solely in industrial and laboratory environments or in the field and must not be used in any way that may cause personal injury or property damage. You are responsible if the product is used for an intention other than its designated purpose or in disregard of the manufacturer's instructions. The manufacturer shall assume no responsibility for such use of the product.

The product is used for its designated purpose if it is used in accordance with its operating manual and within its performance limits (see data sheet, documentation, the following safety instructions). Using the products requires technical skills and knowledge of English. It is therefore essential that the products be used exclusively by skilled and specialized staff or thoroughly trained personnel with the required skills. If personal safety gear is required for using Rohde & Schwarz products, this will be indicated at the appropriate place in the product documentation.

Symbols and safety labels

							
Observe operating instructions	Weight indication for units >18 kg	Danger of electric shock	Warning! Hot surface	PE terminal	Ground	Ground terminal	Attention! Electrostatic sensitive devices

I ○	⏻	⎓	∼	≂	⧈
Supply voltage ON/OFF	Standby indication	Direct current (DC)	Alternating current (AC)	Direct/alternating current (DC/AC)	Device fully protected by double/reinforced insulation

Observing the safety instructions will help prevent personal injury or damage of any kind caused by dangerous situations. Therefore, carefully read through and adhere to the following safety instructions before putting the product into operation. It is also absolutely essential to observe the additional safety instructions on personal safety that appear in other parts of the documentation. In these safety instructions, the word "product" refers to all merchandise sold and distributed by Rohde & Schwarz, including instruments, systems and all accessories.

Tags and their meaning

DANGER	This tag indicates a safety hazard with a high potential of risk for the user that can result in death or serious injuries.
WARNING	This tag indicates a safety hazard with a medium potential of risk for the user that can result in death or serious injuries.
CAUTION	This tag indicates a safety hazard with a low potential of risk for the user that can result in slight or minor injuries.
ATTENTION	This tag indicates the possibility of incorrect use that can cause damage to the product.
NOTE	This tag indicates a situation where the user should pay special attention to operating the product but which does not lead to damage.

These tags are in accordance with the standard definition for civil applications in the European Economic Area. Definitions that deviate from the standard definition may also exist. It is therefore essential to make sure that the tags described here are always used only in connection with the associated documentation and the associated product. The use of tags in connection with unassociated products or unassociated documentation can result in misinterpretations and thus contribute to personal injury or material damage.

Basic safety instructions

1. The product may be operated only under the operating conditions and in the positions specified by the manufacturer. Its ventilation must not be obstructed during operation. Unless otherwise specified, the following requirements apply to Rohde & Schwarz products:
   prescribed operating position is always with the housing floor facing down, IP protection 2X, pollution severity 2, overvoltage category 2, use only in enclosed spaces, max. operation altitude max. 2000 m.
   Unless specified otherwise in the data sheet, a tolerance of ±10% shall apply to the nominal voltage and of ±5% to the nominal frequency.
2. Applicable local or national safety regulations and rules for the prevention of accidents must be observed in all work performed. The product may be opened only by authorized, specially trained personnel. Prior to performing any work on the product or opening the product, the product must be disconnected from the supply network. Any adjustments, replacements of parts, maintenance or repair must be carried out only by technical personnel authorized by Rohde & Schwarz. Only original parts may be used for replacing parts relevant to safety (e.g. power switches, power transformers, fuses). A safety test must always be performed after parts relevant to safety have been replaced (visual inspection, PE conductor test, insulation resistance measurement, leakage current measurement, functional test).
3. As with all industrially manufactured goods, the use of substances that induce an allergic reaction (allergens, e.g. nickel) such as aluminum cannot be generally excluded. If you develop an allergic reaction (such as a skin rash, frequent sneezing, red eyes or respiratory difficulties), consult a physician immediately to determine the cause.

4. If products/components are mechanically and/or thermically processed in a manner that goes beyond their intended use, hazardous substances (heavy-metal dust such as lead, beryllium, nickel) may be released. For this reason, the product may only be disassembled, e.g. for disposal purposes, by specially trained personnel. Improper disassembly may be hazardous to your health. National waste disposal regulations must be observed.

5. If handling the product yields hazardous substances or fuels that must be disposed of in a special way, e.g. coolants or engine oils that must be replenished regularly, the safety instructions of the manufacturer of the hazardous substances or fuels and the applicable regional waste disposal regulations must be observed. Also observe the relevant safety instructions in the product documentation.

6. Depending on the function, certain products such as RF radio equipment can produce an elevated level of electromagnetic radiation. Considering that unborn life requires increased protection, pregnant women should be protected by appropriate measures. Persons with pacemakers may also be endangered by electromagnetic radiation. The employer is required to assess workplaces where there is a special risk of exposure to radiation and, if necessary, take measures to avert the danger.

7. Operating the products requires special training and intense concentration. Make certain that persons who use the products are physically, mentally and emotionally fit enough to handle operating the products; otherwise injuries or material damage may occur. It is the responsibility of the employer to select suitable personnel for operating the products.

8. Prior to switching on the product, it must be ensured that the nominal voltage setting on the product matches the nominal voltage of the AC supply network. If a different voltage is to be set, the power fuse of the product may have to be changed accordingly.

9. In the case of products of safety class I with movable power cord and connector, operation is permitted only on sockets with earthing contact and protective earth connection.

10. Intentionally breaking the protective earth connection either in the feed line or in the product itself is not permitted. Doing so can result in the danger of an electric shock from the product. If extension cords or connector strips are implemented, they must be checked on a regular basis to ensure that they are safe to use.

11. If the product has no power switch for disconnection from the AC supply, the plug of the connecting cable is regarded as the disconnecting device. In such cases, it must be ensured that the power plug is easily reachable and accessible at all times (length of connecting cable approx. 2 m). Functional or electronic switches are not suitable for providing disconnection from the AC supply. If products without power switches are integrated in racks or systems, a disconnecting device must be provided at the system level.

12. Never use the product if the power cable is damaged. By taking appropriate safety measures and carefully laying the power cable, ensure that the cable cannot be damaged and that no one can be hurt by e.g. tripping over the cable or suffering an electric shock.

13. The product may be operated only from TN/TT supply networks fused with max. 16 A.

14. Do not insert the plug into sockets that are dusty or dirty. Insert the plug firmly and all the way into the socket. Otherwise this can result in sparks, fire and/or injuries.

15. Do not overload any sockets, extension cords or connector strips; doing so can cause fire or electric shocks.

16. For measurements in circuits with voltages V_{rms} > 30 V, suitable measures (e.g. appropriate measuring equipment, fusing, current limiting, electrical separation, insulation) should be taken to avoid any hazards.

17. Ensure that the connections with information technology equipment comply with IEC 950/EN 60950.

18. Never remove the cover or part of the housing while you are operating the product. This will expose circuits and components and can lead to injuries, fire or damage to the product.

19. If a product is to be permanently installed, the connection between the PE terminal on site and the product's PE conductor must be made first before any other connection is made. The product may be installed and connected only by a skilled electrician.
20. For permanently installed equipment without built-in fuses, circuit breakers or similar protective devices, the supply circuit must be fused in such a way that suitable protection is provided for users and products.
21. Do not insert any objects into the openings in the housing that are not designed for this purpose. Never pour any liquids onto or into the housing. This can cause short circuits inside the product and/or electric shocks, fire or injuries.
22. Use suitable overvoltage protection to ensure that no overvoltage (such as that caused by a thunderstorm) can reach the product. Otherwise the operating personnel will be endangered by electric shocks.
23. Rohde & Schwarz products are not protected against penetration of water, unless otherwise specified (see also safety instruction 1.). If this is not taken into account, there exists the danger of electric shock or damage to the product, which can also lead to personal injury.
24. Never use the product under conditions in which condensation has formed or can form in or on the product, e.g. if the product was moved from a cold to a warm environment.
25. Do not close any slots or openings on the product, since they are necessary for ventilation and prevent the product from overheating. Do not place the product on soft surfaces such as sofas or rugs or inside a closed housing, unless this is well ventilated.
26. Do not place the product on heat-generating devices such as radiators or fan heaters. The temperature of the environment must not exceed the maximum temperature specified in the data sheet.
27. Batteries and storage batteries must not be exposed to high temperatures or fire. Keep batteries and storage batteries away from children. If batteries or storage batteries are improperly replaced, this can cause an explosion (warning: lithium cells). Replace the battery or storage battery only with the matching Rohde & Schwarz type (see spare parts list). Batteries and storage batteries are hazardous waste. Dispose of them only in specially marked containers. Observe local regulations regarding waste disposal. Do not short-circuit batteries or storage batteries.
28. Please be aware that in the event of a fire, toxic substances (gases, liquids etc.) that may be hazardous to your health may escape from the product.
29. Please be aware of the weight of the product. Be careful when moving it; otherwise you may injure your back or other parts of your body.
30. Do not place the product on surfaces, vehicles, cabinets or tables that for reasons of weight or stability are unsuitable for this purpose. Always follow the manufacturer's installation instructions when installing the product and fastening it to objects or structures (e.g. walls and shelves).
31. Handles on the products are designed exclusively for personnel to hold or carry the product. It is therefore not permissible to use handles for fastening the product to or on means of transport such as cranes, fork lifts, wagons, etc. The user is responsible for securely fastening the products to or on the means of transport and for observing the safety regulations of the manufacturer of the means of transport. Noncompliance can result in personal injury or material damage.
32. If you use the product in a vehicle, it is the sole responsibility of the driver to drive the vehicle safely. Adequately secure the product in the vehicle to prevent injuries or other damage in the event of an accident. Never use the product in a moving vehicle if doing so could distract the driver of the vehicle. The driver is always responsible for the safety of the vehicle; the manufacturer assumes no responsibility for accidents or collisions.
33. If a laser product (e.g. a CD/DVD drive) is integrated in a Rohde & Schwarz product, do not use any other settings or functions than those described in the documentation. Otherwise this may be hazardous to your health, since the laser beam can cause irreversible damage to your eyes. Never try to take such products apart, and never look into the laser beam.

Por favor lea imprescindiblemente antes de la primera puesta en funcionamiento las siguientes informaciones de seguridad

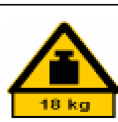
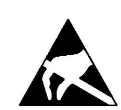
Informaciones de seguridad

Es el principio de Rohde & Schwarz de tener a sus productos siempre al día con los estandards de seguridad y de ofrecer a sus clientes el máximo grado de seguridad. Nuestros productos y todos los equipos adicionales son siempre fabricados y examinados según las normas de seguridad vigentes. Nuestra sección de gestión de la seguridad de calidad controla constantemente que sean cumplidas estas normas. Este producto ha sido fabricado y examinado según el comprobante de conformidad adjunto según las normas de la CE y ha salido de nuestra planta en estado impecable según los estandards técnicos de seguridad. Para poder preservar este estado y garantizar un funcionamiento libre de peligros, deberá el usuario atenerse a todas las informaciones, informaciones de seguridad y notas de alerta. Rohde&Schwarz está siempre a su disposición en caso de que tengan preguntas referentes a estas informaciones de seguridad.

Además queda en la responsabilidad del usuario utilizar el producto en la forma debida. Este producto solamente fue elaborado para ser utilizado en la indústria y el laboratorio o para fines de campo y de ninguna manera deberá ser utilizado de modo que alguna persona/cosa pueda ser dañada. El uso del producto fuera de sus fines definidos o despreciando las informaciones de seguridad del fabricante queda en la responsabilidad del usuario. El fabricante no se hace en ninguna forma responsable de consecuencias a causa del maluso del producto.

Se parte del uso correcto del producto para los fines definidos si el producto es utilizado dentro de las instrucciones del correspondiente manual del uso y dentro del margen de rendimiento definido (ver hoja de datos, documentación, informaciones de seguridad que siguen). El uso de los productos hace necesarios conocimientos profundos y el conocimiento del idioma inglés. Por eso se deberá tener en cuenta de exclusivamente autorizar para el uso de los productos a personas péritas o debidamente minuciosamente instruidas con los conocimientos citados. Si fuera necesaria indumentaria de seguridad para el uso de productos de R&S, encontrará la información debida en la documentación del producto en el capítulo correspondiente.

Símbolos y definiciones de seguridad

							
Ver manual de instrucciones del uso	Informaciones para maquinaria con uns peso de > 18kg	Peligro de golpe de corriente	¡Advertencia! Superficie caliente	Conexión a conductor protector	Conexión a tierra	Conexión a masa conductora	¡Cuidado! Elementos de construción con peligro de carga electroestática

					
potencia EN MARCHA/PARADA	Indicación Stand-by	Corriente continua DC	Corriente alterna AC	Corriente continua/alterna DC/AC	El aparato está protegido en su totalidad por un aislamiento de doble refuerzo

Tener en cuenta las informaciones de seguridad sirve para tratar de evitar daños y peligros de toda clase. Es necesario de que se lean las siguientes informaciones de seguridad concienzudamente y se tengan en cuenta debidamente antes de la puesta en funcionamiento del producto. También deberán ser tenidas en cuenta las informaciones para la protección de personas que encontrarán en otro capítulo de esta documentación y que también son obligatorias de seguir. En las informaciones de seguridad actuales hemos juntado todos los objetos vendidos por Rohde&Schwarz bajo la denominación de „producto“, entre ellos también aparatos, instalaciones así como toda clase de accesorios.

Palabras de señal y su significado

PELIGRO	Indica un punto de peligro con gran potencial de riesgo para el ususario.Punto de peligro que puede llevar hasta la muerte o graves heridas.
ADVERTENCIA	Indica un punto de peligro con un protencial de riesgo mediano para el usuario. Punto de peligro que puede llevar hasta la muerte o graves heridas .
ATENCIÓN	Indica un punto de peligro con un protencial de riesgo pequeño para el usuario. Punto de peligro que puede llevar hasta heridas leves o pequeñas
CUIDADO	Indica la posibilidad de utilizar mal el producto y a consecuencia dañarlo.
INFORMACIÓN	Indica una situación en la que deberían seguirse las instrucciones en el uso del producto, pero que no consecuentemente deben de llevar a un daño del mismo.

Las palabras de señal corresponden a la definición habitual para aplicaciones civiles en el ámbito de la comunidad económica europea. Pueden existir definiciones diferentes a esta definición. Por eso se debera tener en cuenta que las palabras de señal aquí descritas sean utilizadas siempre solamente en combinación con la correspondiente documentación y solamente en combinación con el producto correspondiente. La utilización de las palabras de señal en combinación con productos o documentaciones que no les correspondan puede llevar a malinterpretaciones y tener por consecuencia daños en personas u objetos.

Informaciones de seguridad elementales

1. El producto solamente debe ser utilizado según lo indicado por el fabricante referente a la situación y posición de funcionamiento sin que se obstruya la ventilación. Si no se convino de otra manera, es para los productos R&S válido lo que sigue:
   como posición de funcionamiento se define principialmente la posición con el suelo de la caja para abajo , modo de protección IP 2X, grado de suciedad 2, categoría de sobrecarga eléctrica 2, utilizar solamente en estancias interiores, utilización hasta 2000 m sobre el nivel del mar.
   A menos que se especifique otra cosa en la hoja de datos, se aplicará una tolerancia de ±10% sobre el voltaje nominal y de ±5% sobre la frecuencia nominal.
2. En todos los trabajos deberán ser tenidas en cuenta las normas locales de seguridad de trabajo y de prevención de accidentes. El producto solamente debe de ser abierto por personal périto autorizado. Antes de efectuar trabajos en el producto o abrirlo deberá este ser desconectado de la corriente. El ajuste, el cambio de partes, la manutención y la reparación deberán ser solamente efectuadas por electricistas autorizados por R&S. Si se reponen partes con importancia para los aspectos de seguridad (por ejemplo el enchufe, los transformadores o los fusibles), solamente podrán ser sustituidos por partes originales. Despues de cada recambio de partes elementales para la seguridad deberá ser efectuado un control de

seguridad (control a primera vista, control de conductor protector, medición de resistencia de aislamiento, medición de medición de la corriente conductora, control de funcionamiento).

3. Como en todo producto de fabricación industrial no puede ser excluido en general de que se produzcan al usarlo elementos que puedan generar alergias, los llamados elementos alergénicos (por ejemplo el níquel). Si se producieran en el trato con productos R&S reacciones alérgicas, como por ejemplo urticaria, estornudos frecuentes, irritación de la conjuntiva o dificultades al respirar, se deberá consultar inmediatamente a un médico para averigurar los motivos de estas reacciones.

4. Si productos / elementos de construcción son tratados fuera del funcionamiento definido de forma mecánica o térmica, pueden generarse elementos peligrosos (polvos de sustancia de metales pesados como por ejemplo plomo, berilio, níquel). La partición elemental del producto, como por ejemplo sucede en el tratamiento de materias residuales, debe de ser efectuada solamente por personal especializado para estos tratamientos. La partición elemental efectuada inadecuadamente puede generar daños para la salud. Se deben tener en cuenta las directivas nacionales referentes al tratamiento de materias residuales.

5. En el caso de que se produjeran agentes de peligro o combustibles en la aplicación del producto que debieran de ser transferidos a un tratamiento de materias residuales, como por ejemplo agentes refrigerantes que deben ser repuestos en periodos definidos, o aceites para motores, deberan ser tenidas en cuenta las prescripciones de seguridad del fabricante de estos agentes de peligro o combustibles y las regulaciones regionales para el tratamiento de materias residuales. Cuiden también de tener en cuenta en caso dado las prescripciones de seguridad especiales en la descripción del producto.

6. Ciertos productos, como por ejemplo las instalaciones de radiación HF, pueden a causa de su función natural, emitir una radiación electromagnética aumentada. En vista a la protección de la vida en desarrollo deberían ser protegidas personas embarazadas debidamente. También las personas con un bypass pueden correr peligro a causa de la radiación electromagnética. El empresario está comprometido a valorar y señalar areas de trabajo en las que se corra un riesgo de exposición a radiaciones aumentadas de riesgo aumentado para evitar riesgos.

7. La utilización de los productos requiere instrucciones especiales y una alta concentración en el manejo. Debe de ponerse por seguro de que las personas que manejen los productos estén a la altura de los requerimientos necesarios referente a sus aptitudes físicas, psíquicas y emocionales, ya que de otra manera no se pueden excluir lesiones o daños de objetos. El empresario lleva la responsabilidad de seleccionar el personal usuario apto para el manejo de los productos.

8. Antes de la puesta en marcha del producto se deberá tener por seguro de que la tensión preseleccionada en el producto equivalga a la del la red de distribución. Si es necesario cambiar la preselección de la tensión también se deberán en caso dabo cambiar los fusibles correspondientes del prodcuto.

9. Productos de la clase de seguridad I con alimentación móvil y enchufe individual de producto solamente deberán ser conectados para el funcionamiento a tomas de corriente de contacto de seguridad y con conductor protector conectado.

10. Queda prohibida toda clase de interrupción intencionada del conductor protector, tanto en la toma de corriente como en el mismo producto ya que puede tener como consecuencia el peligro de golpe de corriente por el producto. Si se utilizaran cables o enchufes de extensión se deberá poner al seguro, que es controlado su estado técnico de seguridad.

11. Si el producto no está equipado con un interruptor para desconectarlo de la red, se deberá considerar el enchufe del cable de distribución como interruptor. En estos casos deberá asegurar de que el enchufe sea de fácil acceso y nabejo (medida del cable de distribución aproximadamente 2 m). Los interruptores de función o electrónicos no son aptos para el corte de la red eléctrica. Si los productos sin interruptor están integrados en construciones o instalaciones, se deberá instalar el interruptor al nivel de la instalación.

12. No utilice nunca el producto si está dañado el cable eléctrico. Asegure a través de las medidas de protección y de instalación adecuadas de que el cable de eléctrico no pueda ser dañado o de que nadie pueda ser dañado por él, por ejemplo al tropezar o por un golpe de corriente.
13. Solamente está permitido el funcionamiento en redes de distribución TN/TT aseguradas con fusibles de como máximo 16 A.
14. Nunca conecte el enchufe en tomas de corriente sucias o llenas de polvo. Introduzca el enchufe por completo y fuertemente en la toma de corriente. Si no tiene en consideración estas indicaciones se arriesga a que se originen chispas, fuego y/o heridas.
15. No sobrecargue las tomas de corriente, los cables de extensión o los enchufes de extensión ya que esto pudiera causar fuego o golpes de corriente.
16. En las mediciones en circuitos de corriente con una tensión de entrada de Ueff > 30 V se deberá tomar las precauciones debidas para impedir cualquier peligro (por ejemplo medios de medición adecuados, seguros, limitación de tensión, corte protector, aislamiento etc.).
17. En caso de conexión con aparatos de la técnica informática se deberá tener en cuenta que estos cumplan los requisitos de la EC950/EN60950.
18. Nunca abra la tapa o parte de ella si el producto está en funcionamiento. Esto pone a descubierto los cables y componentes eléctricos y puede causar heridas, fuego o daños en el producto.
19. Si un producto es instalado fijamente en un lugar, se deberá primero conectar el conductor protector fijo con el conductor protector del aparato antes de hacer cualquier otra conexión. La instalación y la conexión deberán ser efecutadas por un electricista especializado.
20. En caso de que los productos que son instalados fijamente en un lugar sean sin protector implementado, autointerruptor o similares objetos de protección, deberá la toma de corriente estar protegida de manera que los productos o los usuarios estén suficientemente protegidos.
21. Por favor, no introduzca ningún objeto que no esté destinado a ello en los orificios de la caja del aparato. No vierta nunca ninguna clase de líquidos sobre o en la caja. Esto puede producir corto circuitos en el producto y/o puede causar golpes de corriente, fuego o heridas.
22. Asegúrese con la protección adecuada de que no pueda originarse en el producto una sobrecarga por ejemplo a causa de una tormenta. Si no se verá el personal que lo utilice expuesto al peligro de un golpe de corriente.
23. Los productos R&S no están protegidos contra el agua si no es que exista otra indicación, ver también punto 1. Si no se tiene en cuenta esto se arriesga el peligro de golpe de corriente o de daños en el producto lo cual también puede llevar al peligro de personas.
24. No utilice el producto bajo condiciones en las que pueda producirse y se hayan producido líquidos de condensación en o dentro del producto como por ejemplo cuando se desplaza el producto de un lugar frío a un lugar caliente.
25. Por favor no cierre ninguna ranura u orificio del producto, ya que estas son necesarias para la ventilación e impiden que el producto se caliente demasiado. No pongan el producto encima de materiales blandos como por ejemplo sofás o alfombras o dentro de una caja cerrada, si esta no está suficientemente ventilada.
26. No ponga el producto sobre aparatos que produzcan calor, como por ejemplo radiadores o calentadores. La temperatura ambiental no debe superar la temperatura máxima especificada en la hoja de datos.

27. Baterías y acumuladores no deben de ser expuestos a temperaturas altas o al fuego. Guardar baterías y acumuladores fuera del alcance de los niños. Si las baterías o los acumuladores no son cambiados con la debida atención existirá peligro de explosión (atención celulas de Litio). Cambiar las baterías o los acumuladores solamente por los del tipo R&S correspondiente (ver lista de piezas de recambio). Baterías y acumuladores son deshechos problemáticos. Por favor tirenlos en los recipientes especiales para este fín. Por favor tengan en cuenta las prescripciones nacionales de cada país referente al tratamiento de deshechos. Nunca sometan las baterías o acumuladores a un corto circuito.

28. Tengan en consideración de que en caso de un incendio pueden escaparse gases tóxicos del producto, que pueden causar daños a la salud.

29. Por favor tengan en cuenta que en caso de un incendio pueden desprenderse del producto agentes venenosos (gases, líquidos etc.) que pueden generar daños a la salud.

30. No sitúe el producto encima de superficies, vehículos, estantes o mesas, que por sus características de peso o de estabilidad no sean aptas para él. Siga siempre las instrucciones de instalación del fabricante cuando instale y asegure el producto en objetos o estructuras (por ejemplo paredes y estantes).

31. Las asas instaladas en los productos sirven solamente de ayuda para el manejo que solamente está previsto para personas. Por eso no está permitido utilizar las asas para la sujecion en o sobre medios de transporte como por ejemplo grúas, carretillas elevadoras de horquilla, carros etc. El usuario es responsable de que los productos sean sujetados de forma segura a los medios de transporte y de que las prescripciones de seguridad del fabricante de los medios de transporte sean tenidas en cuenta. En caso de que no se tengan en cuenta pueden causarse daños en personas y objetos.

32. Si llega a utilizar el producto dentro de un vehículo, queda en la responsabilidad absoluta del conductor que conducir el vehículo de manera segura. Asegure el producto dentro del vehículo debidamente para evitar en caso de un accidente las lesiones u otra clase de daños. No utilice nunca el producto dentro de un vehículo en movimiento si esto pudiera distraer al conductor. Siempre queda en la responsabilidad absoluta del conductor la seguridad del vehículo y el fabricante no asumirá ninguna clase de responsabilidad por accidentes o colisiones.

33. Dado el caso de que esté integrado un producto de laser en un producto R&S (por ejemplo CD/DVD-ROM) no utilice otras instalaciones o funciones que las descritas en la documentación. De otra manera pondrá en peligro su salud, ya que el rayo laser puede dañar irreversiblemente sus ojos. Nunca trate de descomponer estos productos. Nunca mire dentro del rayo laser.

Certified Quality System

DIN EN ISO 9001 : 2000
DIN EN 9100 : 2003
DIN EN ISO 14001 : 1996

DQS REG. NO 001954 QM/ST UM

QUALITÄTSZERTIFIKAT

Sehr geehrter Kunde,
Sie haben sich für den Kauf eines Rohde & Schwarz-Produktes entschieden. Hiermit erhalten Sie ein nach modernsten Fertigungsmethoden hergestelltes Produkt. Es wurde nach den Regeln unseres Managementsystems entwickelt, gefertigt und geprüft.
Das Rohde & Schwarz Managementsystem ist zertifiziert nach:

DIN EN ISO 9001:2000
DIN EN 9100:2003
DIN EN ISO 14001:1996

CERTIFICATE OF QUALITY

Dear Customer,
you have decided to buy a Rohde & Schwarz product. You are thus assured of receiving a product that is manufactured using the most modern methods available. This product was developed, manufactured and tested in compliance with our quality management system standards.
The Rohde & Schwarz quality management system is certified according to:

DIN EN ISO 9001:2000
DIN EN 9100:2003
DIN EN ISO 14001:1996

CERTIFICAT DE QUALITÉ

Cher Client,
vous avez choisi d'acheter un produit Rohde & Schwarz. Vous disposez donc d'un produit fabriqué d'après les méthodes les plus avancées. Le développement, la fabrication et les tests respectent nos normes de gestion qualité.
Le système de gestion qualité de Rohde & Schwarz a été homologué conformément aux normes:

DIN EN ISO 9001:2000
DIN EN 9100:2003
DIN EN ISO 14001:1996

ROHDE&SCHWARZ

Customer Support

Technical support – where and when you need it

For quick, expert help with any Rohde & Schwarz equipment, contact one of our Customer Support Centers. A team of highly qualified engineers provides telephone support and will work with you to find a solution to your query on any aspect of the operation, programming or applications of Rohde & Schwarz equipment.

Up-to-date information and upgrades

To keep your Rohde & Schwarz equipment always up-to-date, please subscribe to our electronic newsletter at

http://www.rohde-schwarz.com/www/response.nsf/newsletterpreselection

or request the desired information and upgrades via email from your Customer Support Center (addresses see below).

Feedback

We want to know if we are meeting your support needs. If you have any comments please email us and let us know CustomerSupport.Feedback@rohde-schwarz.com.

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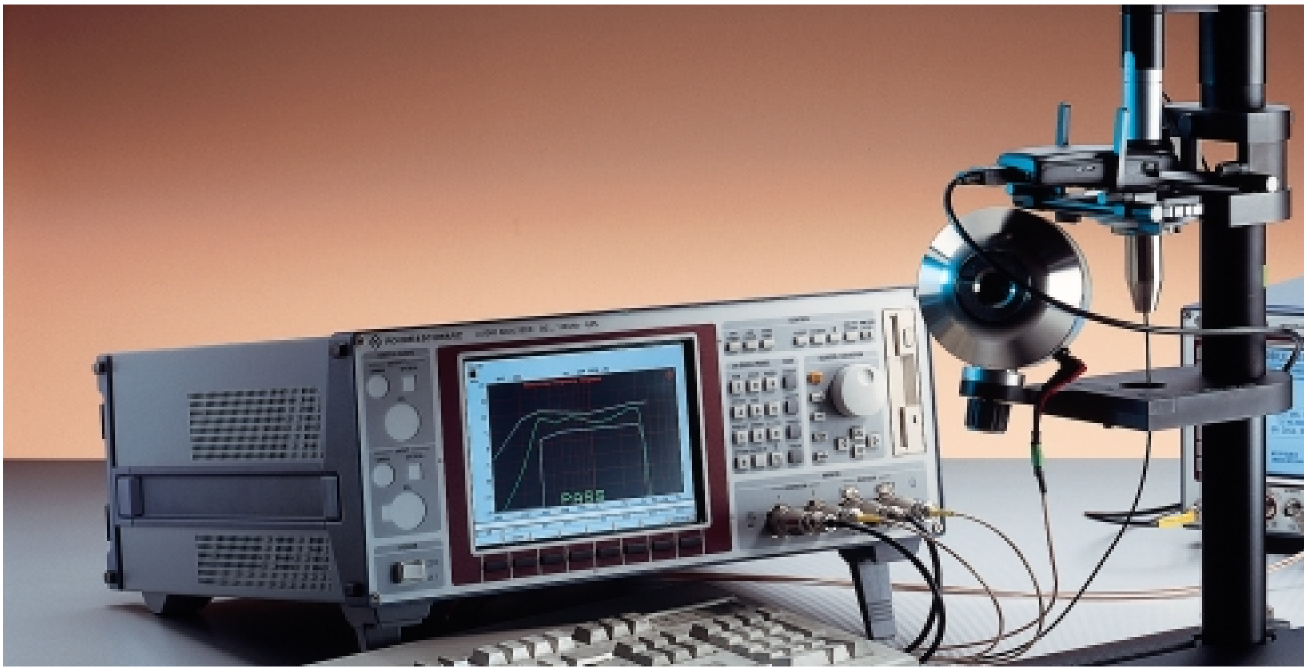
Technical Information



Audio Analyzer UPL16

Acoustic Tests of GSM Mobiles

The latest member in the family of Audio Analyzers UPL, the new model UPL16 is especially designed for acoustic tests of GSM mobiles. Therefore the instrument is equipped with the Digital Audio Interface DAI, artificial voice generator and the necessary software routines. Together with the Digital Radiocommunication Testers CRTx or CMU, and completed with an artificial mouth/ear combination, the UPL16 provides a complete system handling the acoustic testcases according to 3GPP TS 51.010-1, section 30, up to release 99.

- Digital Audio Interface DAI built-in
- Test cases according to 3GPP TS 51.010-1
- All kinds of analog and digital test signals including artificial voice
- Measurements in analog and digital domain
- UPL16 is part of Type Approval System TS8916
- UPL16 + CRTx or CMU for type approval measurements
- Validated Measurements via Air Interface with Option UPL-B9

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Brief description

The Audio Analyzer UPL16 offers the same superior analysis concept as all members of the UPL family. The instrument is able to handle both analog and digital processing, it performs practically all types of measurements in the audio range from frequency response measurements through distortion measurements, FFT analysis, etc. All the details about test functions and the concept of operation can be found in the data sheet of the Audio Analyzer UPL.
The UPL16 is especially designed for acoustic tests of GSM mobiles. It is equipped with the Digital Audio Interface DAI as defined in 3GPP TS 44.014. With this interface the UPL16 can act as a system simulator or it can simulate a mobile.
To drive an artificial head directly without the need of additional amplifiers, the generator output of the UPL16 is matched to the impedance of the artificial mouth by a built-in transformer.
3GPP TS 51.010-1, section 30 defines so called speech teleservices tests for GSM mobiles, generally referred in this document as acoustic testcases. All measurements via DAI are supported by the software package delivered with the instrument. The acoustic tests via Air Interface are available with option UPL-B9.

UPL16 in test system applications

The Audio Analyzer UPL16 can be used in system applications in different ways:

UPL16 together with Digital Radiocommunication Tester CRTx

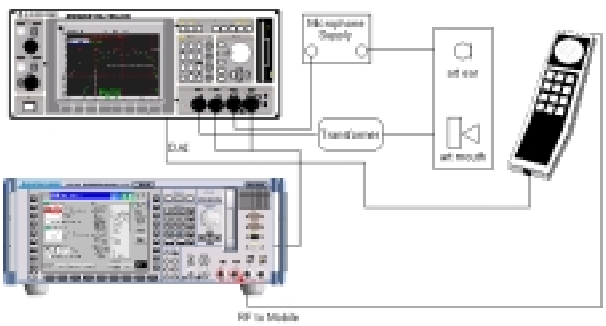
This application is ideal for type approval and development labs. For CRTx the software package CRTKAUD is required. The test cases are started from the user interface of the CRTx, the UPL16 is controlled via IEEE-Bus. The results are displayed on the UPL16 and transferred to CRTx. Together with the UPL16 acoustic measurements are now added to the large variety of tests provided by CRTx.

UPL16 as part of GSM Simulator TS8916

The application UPL16 with CRTx is fully integrated in the TS8916 system. Performance and operation concerning the control of UPL, activating testcases and speech coder via CRTKAUD, as well as the result handling are practically identical on CRTx and TS8916.

UPL16 together with Universal Radiocommunication Tester CMU200

This is the ideal and cost-effective combination for testing the acoustic quality in type approval, development, quality assurance, production, etc.
The CMU sets up a call to the mobile via the RF link. Then the test cases are started from the user interface of the UPL16, the results are displayed on the audio analyzer. For some of the tests, the speech codec is needed, which is implemented in the CMU200 with option CMU-B52.



Audio Analyzer UPL16 provides all the necessary audio signals and performs the acoustic measurements. Universal Radiocommunication Tester CMU is used to establish the call. For some test cases also the speech codec is used.

Acoustic test cases according to 3GPP TS 51.010-1 section 30

Acoustic Tests of GSM and 3GPP Mobile Phones	**TS 51.010 GSM Rel. 99** DAI interface Chapter	**TS 51.010 GSM Rel. 4** Air interface Chapter
Sending frequency response	**30.1***	**30.12***
Sending loudness rating	**30.2***	**30.13***
Receiving freq. response	**30.3***	**30.14***
Receiving loudness rating	**30.4***	**30.15***
Side tone masking rating	**30.5.1***	**30.16***
Listener side tone rating	30.5.2	
Echo loss	30.6.1 (rel. 4)	**30.17.1***
Stability margin	**30.6.2***	**30.17.2***
Sending distortion	**30.7.1***	**30.18***
Receiving distortion	30.7.2	
Sidetone distortion	30.8	
Out-of-band signals	30.9	
Idle channel noise	30.10	
Ambient noise rejection	30.11 (rel. 4)	**30.19***

*** validated testcases**

Air interface tests available with option UPL-B9

Listener Sidetone Rating and Ambient Noise Rejection require additional hardware for sound field generation.

DAI operating modes

Interface DAI to MS
This is the normal operating mode which should always be used, when connecting the UPL to a DAI mobile to perform the acoustical testcases.
The UPL is part of the System Simulator (SS).

Interface DAI to SS
The UPL simulates a mobile (MS) and thus can be connected to the DAI of a System Simulator (SS). This operating mode can be used to test the DAI interface of a SS.

Interface AUX to SS
A second (auxiliary) DAI interface is implemented. Reserved for future use. Identical functions as DAI to SS.

Options for UPL16

The following UPL options are included in UPL16:

- Remote Control UPL-B4
- Extended Analysis Functions UPL-B6
- Universal Sequence Controller UPL-B10

Digital Audio I/O UPL-B2 or UPL-B29 cannot be used with UPL16.

The options which can additionally be ordered with UPL16 are:

- Non standardised Mobile Phone Measurements via air interface UPL-B8
- Validated Mobile Phone Tests UPL-B9 for air interface tests according to 3GPP TS 26.132
- Low Distortion Generator UPL-B1,
- Audio Monitor UPL-B5 (for analog input only)

Options for CMU200

- Signalling Unit CMU-B21
- Speech Codec CMU-B52
- Software depending on GSM band CMU-K20 ... CMU-K24

Acoustic equipment

Existing equipment can be kept in use. For new installations the following Brüel & Kjaer parts are recommended:

Telephone test head	B&K 4602B
Ear simulator	B&K 4185 Type 1 B&K 4195 Type 3.2 Low leakage and high leakage
Artificial mouth	B&K 4227
Head and torso simulator	B&K 4128C (ear type 3.3)
Acoustic calibrator	B&K 4231
Microphone power supply	B&K 2690A0S2
Acoustic test chamber	e.g. Studio Box Type S

Additional specifications for UPL16

For complete specifications see data sheet UPL (PD 757.2238.24)

Digital audio interface DAI

Implemented according to 3GPP TS 44.014

Connector	25-pole DSUB, male, rear panel
Audio data	13 Bit, linear PCM or via A-Law codec
Input logic state	TTL
Output logic state	Low < 0.8 V, High > 3.5 V
Load impedance	≥ 2 kΩ
Reset pulse	
Duration	5 ms
Rise/fall time	0.5 µs

Test setup for stability margin

Generator function RANDOM+ANLR for digital loop including selectable gain and additional noise signal

Output amplitude (peak)	14 V max
Loop Gain	-240 to +60 dB, selectable
Noise	
Frequency Range	350 Hz to 550 Hz, selectable
Spacing	adjustable from 2.93 Hz
Crest Factor	selectable (3.35 for O.131 Noise)
Delay	1.2 ms

1/3rd octave analysis

No of 1/3rd octaves	30
Frequency Range	22 Hz to 22 kHz
Level accuracy	
Center frequency	± 0.2 dB
22 Hz to 22 kHz	± 1.0 dB (IEC 1260, class 0)

Artificial Voice

male and female according to ITU-T P.50

Audio transformer

Matches the UPL generator output to the impedance of the artificial mouth (loudspeaker).

Connectors	BNC, rear panel
Frequency Range	100 Hz to 10 kHz
Input level (rms)	10 V max
Turns ratio	4:1
Output level (rms)	2.5 V max, open circuit
Output impedance	1.1 Ω
Load impedance	≥ 4 Ω
Frequency response	± 0.5 dB
THD	0.3 %

Ordering information

Audio Analyzer	**UPL16**	1078.2008.16

Options / recommended extras

Low Distortion Generator	UPL-B1	1078.4400.02
Audio Monitor	UPL-B5	1078.4600.03
Mobile Phone Test Set	UPL-B8	1117.3505.02
3G Mobile Phone Tests	UPL-B9	1154.7500.02
XLR/BNC Adapter Set	UPL-Z1	1078.3704.02
19" Rack Adapter	ZZA-94	0396.4905.00

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02.2002

ROHDE&SCHWARZ
EC Certificate of Conformity

CE

Certificate No.: 9502140

This is to certify that:

Equipment type	Order No.	Designation
UPL	1078.2008.02/.05/.06	Audio Analyzer
UPL16	1078.2008.16	
UPL66	1078.2008.66	
UPL-B1	1078.4400.02	Low Distortion Generator
UPL-B2	1078.4000.02	Digital Audio I/O
UPL-B29	1078.5107.02	Digital Audio I/O 96 kHz
UPL-B5	1078.4600.02/.03	Monitoroutput

complies with the provisions of the Directive of the Council of the European Union on the approximation of the laws of the Member States

- relating to electrical equipment for use within defined voltage limits (73/23/EEC revised by 93/68/EEC)
- relating to electromagnetic compatibility (89/336/EEC revised by 91/263/EEC, 92/31/EEC, 93/68/EEC)

Conformity is proven by compliance with the following standards:

EN61010-1 : 1993 + A2 : 1995
EN50081-1 : 1992
EN50082-1 : 1992

Affixing the EC conformity mark as from 1995

ROHDE & SCHWARZ GmbH & Co. KG
Mühldorfstr. 15, D-81671 München

Munich, 1999-03-04

Central Quality Management FS-QZ / Becker

1 Introduction

Audio Analyzer UPL 16 permits acoustic measurements to be performed on GSM mobile stations like those prescribed for the type approval test in line with 3GPP TS 51.010.

The acoustic test cases described in chapter 30, Speech Teleservices, of 3GPP TS 51.010 are measurement procedures with the aid of which specific characteristics of the mobile can be measured.

A GSM system simulator like CRTx, CMD or CMU is required for call setup to the mobile and for executing the test cases 30.6.1 (Echo Loss) and 30.6.2 (Stability Margin) which use the speech codec.

1.1 General

This is a supplementary manual and contains only special cases and extensions offered by UPL 16 as against the basic model UPL 06. Characteristics identical to those of UPL 06 are described in the UPL manual.

A distinction has to be made between references to the supplementary manual and to the UPL operating manual. The latter is referenced to with "see UPL Manual ...".

No changes have been made to the analog section; in the digital section the reduced sample rate limits the frequency range to 4 kHz. There are some differences in the operation of the digital section of generator and analyzer, and in the analyzer this particularly concerns the configuration section. Settings for the digital interface of generator and analyzer are identical and made in the OPTIONS panel.

UPL 16 comprises the following extensions as against the UPL standard model:
The instrument is equipped with a GSM DAI interface according to 3GPP TS 44.014 section 10 and an audio transformer for a direct drive of the artificial mouth. Options UPL-B4, B6 and B10 are standard equipment. The software for acoustic measurements via DAI interface in line with 3GPP TS 51.010 is installed. For acoustic measurements via air interface options UPL-B8 Mobile Phone Test Set and UPL-B9 3G Mobile Phone Tests can be installed.

UPL16 may be equipped in addition with options UPL-B1 and UPL-B5. Options UPL-B2, B21, B22, B23 and B29 cannot be integrated.

1.2 Software Installation

The operating system and the complete UPL software with example files are installed on the hard disk of each UPL supplied. In addition on UPL16 the two software packages U8 (acoustic testcases with CRTx or CMD) and U81 (acoustic testcases with CMU) are installed. The supplied floppy disks are needed only when the complete software or parts thereof have been deleted accidentally by the user, if certain files are corrupted or when the hard disk is to be changed. The following floppy disks are supplied together with the UPL:

- MS-DOS system floppies, containing all programs associated with MS-DOS.
- UPL program floppies, including the complete UPL operating and measurement software.
- UPL example disk. It contains examples for remote control via IEC/IEEE-bus and for Universal Sequence Controller UPL-B10 as well as setups to different measurement applications.
- UPL-U8 Software. It contains the BASIC programs and setups for the GSM acoustic test cases using DAI, as well as female and male artificial voices according to ITU-T Recommendation P.50.
- UPL-U81 Software. It contains the BASIC programs and setups for the GSM acoustic test cases release 99 with CMU

Each of the mentioned program packages can also be installed separately.

Note: *The UPL software is supplied in packed format and unpacked only during installation (the software then considerably exceeds the capacity available on the disk). The unpacking program may output messages such as "Exploding...", "Unpacking" etc. These messages are correct and do not mean faulty installation.*

Installing the MS-DOS operating system:

- Connect the external keyboard.
- Insert installation disk #1.
- Switch on UPL.
- Enter A:SETUP
- If the UPL software starts up: Exit the UPL operating software by pressing the ESC key while the switch-on logo is being displayed on the screen, or, with the UPL operating software loaded, by pressing the SYSTEM key and Enter (corresponds to "Normal Exit to DOS" in the selection box).
- The installation program is started.

Continue the installation following the notes on the screen.

Installing the UPL operating and measurement software:

- Connect the external keyboard.
- Switch on UPL.
- Exit the UPL operating software by pressing the ESC key while the switch-on logo is being displayed on the screen, or, with the UPL operating software loaded, by pressing the SYSTEM key and Enter (corresponds to "Normal Exit to DOS" in the selection box).
- Insert the UPL program disk.
- Key in A :, press Enter.
- Key in UPLINST, press Enter.

The UPL software is now copied onto the hard disk.
Continue the installation following the notes on the screen.

The UPL user interface is displayed on the screen.

Note: *If an updated version of MS-DOS or of the UPL software is to be installed, proceed as described above.*

Installation of UPL example files: :

- Connect the external keyboard.
- Switch on UPL.
- Exit the UPL operating software by pressing ESC key while the switch-on logo is being displayed on the screen, or, with the UPL operating software loaded, by pressing the SYSTEM key and Enter (corresponds to ”Normal Exit to DOS” in the selection box).
- Insert the UPL example disk.
- Key in A :, press Enter.
- Key in SETINST, press Enter.

The UPL example files are now copied onto the hard disk. Then the UPL operating software can be started as usual.

Installation of U8 software (GSM testcases and Artificial Voice):

- Connect the external keyboard.
- Switch on UPL.
- Exit the UPL operating software by pressing ESC key while the switch-on logo is being displayed on the screen, or, with the UPL operating software loaded, by pressing the SYSTEM key and Enter (corresponds to ”Normal Exit to DOS” in the selection box).
- Insert disk 1 (Artificial Voice female)
- Key in A :, press Enter.
- Key in INSTALL, press Enter.
- Insert disk 2 on request and press any key
- Insert disk 3 on request and press any key
- Key in C:\UPL to start the UPL operating software

The INSTALL program creates the directory C:\GSM (if not already existing) on Audio Analyzer UPL and copies the BASIC program, artificial voice and all setups and files required for that application into this directory.

Installation of U81 software (GSM Testcases Rel-99 with CMU)

- See Annex C

Directory structure on hard disk:

The files copied onto the hard disk during installation are stored in the following directory structure:

```
C:\ ──┬── DOS
      ├── UTIL
      ├── TEMP
      ├── GSM
      ├── MOUSE
      └── UPL ──┬── REF
                ├── SETUP
                ├── USER
                ├── DSP
                ├── DRIVER
                ├── IEC_EXAM
                └── B10_EXAM
```

Fig. 1 Directory structure

The READ.ME file in the C:\UPL directory refers to the contents of the individual directories and files.

Note: *To ensure correct functioning of the UPL measurement and operating software, do not modify the directory structure stated above nor the paths.*

2 Hardware Description

2.1 GSM Digital Audio Interface

The demands set on the GSM digital audio interface (DAI) are defined in 3GPP TS 44.014 specs, chapter 10. The implementation in the UPL fully complies with this specification.

A second (auxiliary) DAI interface is available at unassigned pins.

The EMMI (electrical man machine interface) function is not implemented.

2.1.1 Logic Function

The data exchanged on the interface are 13-bit linear PCM at 8 k samples per second, which are transferred to and from the mobile on two serial lines at 104 kbit/s (13 bit ∗ 8 kHz = 104 kbit/s).
Another line, controlled by the MS, clocks the data.
One additional line, controlled by the SS, resets the speech transcoder and the speech A/D and D/A functions and starts data transmission.

Except for this interface, which is defined by 3GPP TS 44.014 and called "Interface DAI to MS" in the UPL, two other variants are implemented in UPL. The following interfaces are therefore available:

Interface DAI to MS	The UPL acts as a system simulator and can be connected to a mobile.
Interface DAI to SS	The UPL simulates a mobile and can be connected to the DAI of a system simulator (eg for test purposes).
Interface AUX to SS	Second (auxiliary) DAI interface. It has the same function but another pin assignment as the DAI to SS interface. Only the pins defined "not used" by 3GPP TS 44.014 are employed.

Only one of the two interfaces (DAI or AUX) can be used at a time.

Two test control lines determine the operating mode of the mobile. They are controlled by the SS.
Having set the appropriate test control signals a reset pulse has to be applied to activate the test mode in the mobile.

Table 1: test control lines:

Test control line		Function
1	2	
Low	Low	Normal operation (idle mode)
Low	High	Test of speech decoder / DTX functions (downlink)
High	Low	Test of speech encoder / DTX functions (uplink)
High	High	Test of acoustic devices and A/D & D/A

2.1.2 Pin Assignment and Signal Flow

The connector for the GSM digital audio interface is placed at the rear panel. A 25-pin sub-D socket (male) labelled GSM DAI is used.

A second (auxiliary) DAI interface is implemented in addition to that defined by 3GPP TS 44.014. It has the same function as the DAI to SS interface. Only one interface can be used at a time.

This auxiliary interface uses only pins which are defined as "not used" in 3GPP TS 44.014.

The pin assignment of the connector and the signal flow at the interface are shown in the two tables below for the two interfaces:

Interface DAI to MS

Table 2: Pin assignment of GSM digital audio interface:

Pin	Use	Function	Signal flow
1		Ground	
2-3	Not used	EMMI signals	
4	Not used		
5	Aux DAI	Reset	Z
6	Aux DAI	Data	Z
7		Ground	
8-9	Not used		
10		Ground	
11	DAI	Test control 1	UPL → MS
12		Ground	
13	DAI	Test control 2	UPL → MS
14-19	Not used		
20	Aux DAI	Data clock	Z
21	Aux DAI	Data	Z
22	DAI	Reset	UPL → MS
23	DAI	Data	MS → UPL
24	DAI	Data clock	MS → UPL
25	DAI	Data	UPL → MS

Z Not available, lines are high impedance.
↔ Direction of signal flow selectable.
SS System simulator
MS Mobile station

Interface DAI to SS or AUX to SS

Table 3: Pin assignment of GSM digital audio interface:

Pin	Use	Function	Signal flow			
1		Ground				
2-3	Not used	EMMI signals				
4	Not used					
5	Aux DAI	Reset	Z	Z	SS → UPL	SS ↔ UPL
6	Aux DAI	Data	Z	Z	UPL → SS	UPL → SS
7		Ground				
8-9	Not used					
10		Ground				
11	DAI	Test control 1	Z	Z	Z	Z
12		Ground				
13	DAI	Test control 2	Z	Z	Z	Z
14-19	Not used					
20	Aux DAI	Data clock	Z	Z	UPL → SS	UPL ↔ SS
21	Aux DAI	Data	Z	Z	SS → UPL	SS → UPL
22	DAI	Reset	SS → UPL	SS ↔ UPL	Z	Z
23	DAI	Data	UPL → SS	UPL → SS	Z	Z
24	DAI	Data clock	UPL → SS	UPL ↔ SS	Z	Z
25	DAI	Data	SS → UPL	SS → UPL	Z	Z
			Normal	Special	Normal	Special
			DAI to SS		AUX to SS	

Z Not available, lines are high impedance.
↔ Direction of signal flow selectable.
SS System simulator

2.1.3 Timing

Data are read in by the MS or SS at the rising edge and are output by the SS or MS at the falling edge of the clock, as shown in the figure. The data consists of 13 bit words in two's complement format, with the most significant bit transmitted first.

Upon release of the reset pulse the data transmission starts with the first clock pulse. The delay between reset pulse and start of data transmission is not defined.

The reset signal is active low. This state has to be longer than 4 ms. The reset signal always remains high when no reset is sent.

The clock signal is high when no clock is sent.

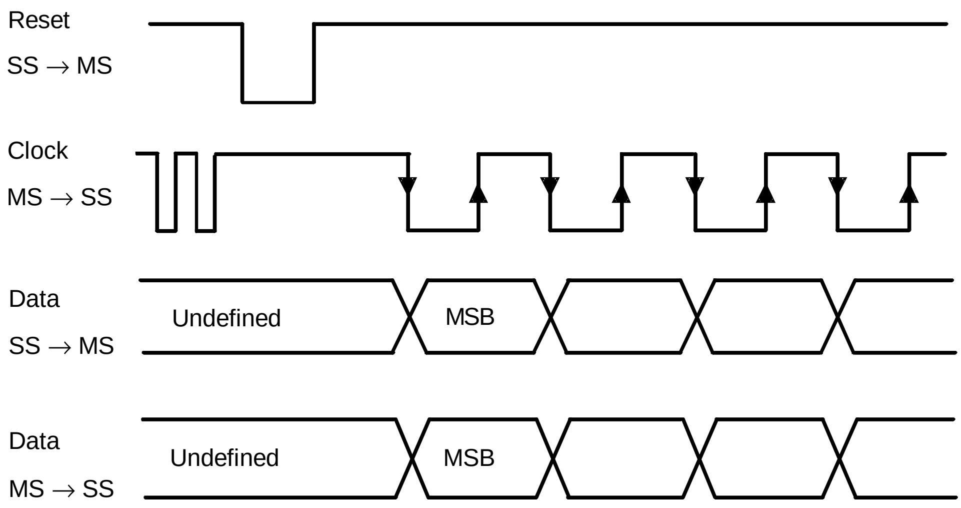


Fig. 2 DAI Timing

2.2 Signal Source for Artificial Mouth

A audio transformer adapts the UPL generator output to the impedance of the artificial mouth (loudspeaker). The transformation ratio is 4:1. Thus a maximum output voltage of 2.5 V is thus available in the idle mode. The output impedance is typically 1 Ω so that a maximum power of approx. 1 W can be output.

At the maximum input voltage of 10 V (rms) a frequency range from 100 Hz to beyond 10 kHz can be used. If the transformer is overloaded and gets into a saturated state, the generator output is switched off. This state is signalled on the UPL by the OUTPUT OFF and GEN OVLD LEDs and the message OUTPUT OFF in the status line of the generator.

BNC connectors are used for input and output. They are located are at the rear of the unit and labelled 4:1 Trans. . The generator output is connected to the transformer input, the output of the transformer to the artificial mouth.

2.3 Test Setup for Stability Margin Measurement

The following test setup is prescribed for test case 30.6.2 (Stability Margin) in 3GPP TS 51.010: the analog output signal of the speech decoder is amplified by 6 dB. A narrowband noise signal in line with ITU-T Recommendation O.131 is added to this signal. The resulting signal is applied to the analog input of the speech decoder. An amplification of the signal at the input or output of the speech decoder must also be considered when the loop gain of 6 dB is adjusted.

The whole loop between output and input of the speech coder including the user-selectable gain and addition of noise are digitally implemented in the UPL. The generator function RANDOM+ANLR is used for this purpose.

The signal from the speech decoder is applied to the analyzer. It is multiplied in the UPL by a gain factor and then the noise signal is added. The signal thus obtained is the generator output signal which is applied to the input of the speech coder.

The signal gain measured by the analyzer and the amplitude of the noise signal are user selectable. The measurement range of the analyzer and the maximum generator output level are to be selected so that the signal is not overdriven in the analyzer or the generator.

3 Measurement Equipment

3.1 UPL with CRTx

UPL is controlled by a Radio Communication Tester CRTx via IEC/IEEE bus. CRTx sets up the communication with the mobile. Any test case can then be performed. The measurement sequences are in the form of BASIC programs which run on the UPL. Measurement results are forwarded to the CRTx for evaluation.

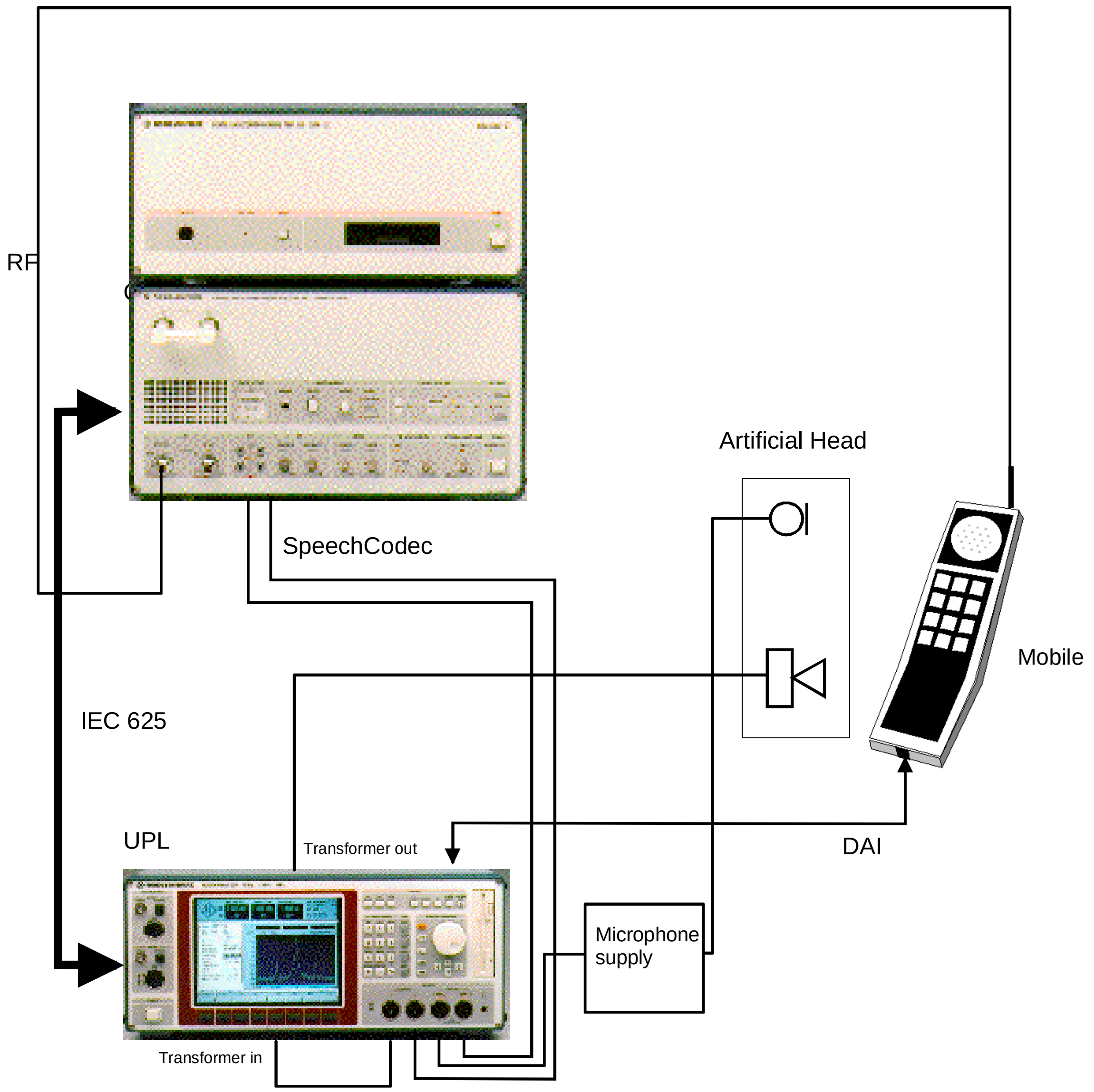


Fig. 3 Test setup UPL with CRTx

3.2 UPL with CMD

CMD sets up communication with the mobile. The test cases are started from the UPL. The measurement sequences are in the form of BASIC programs which run on the UPL.
The measurement results are displayed on the UPL screen. CMU200 can be used instead of CMD.
A special sequence controller is available for calling up the test cases and for result display (see Annex B). When using this test setup the testcases are not validated.
Calibration procedure is described in chapter 3.6

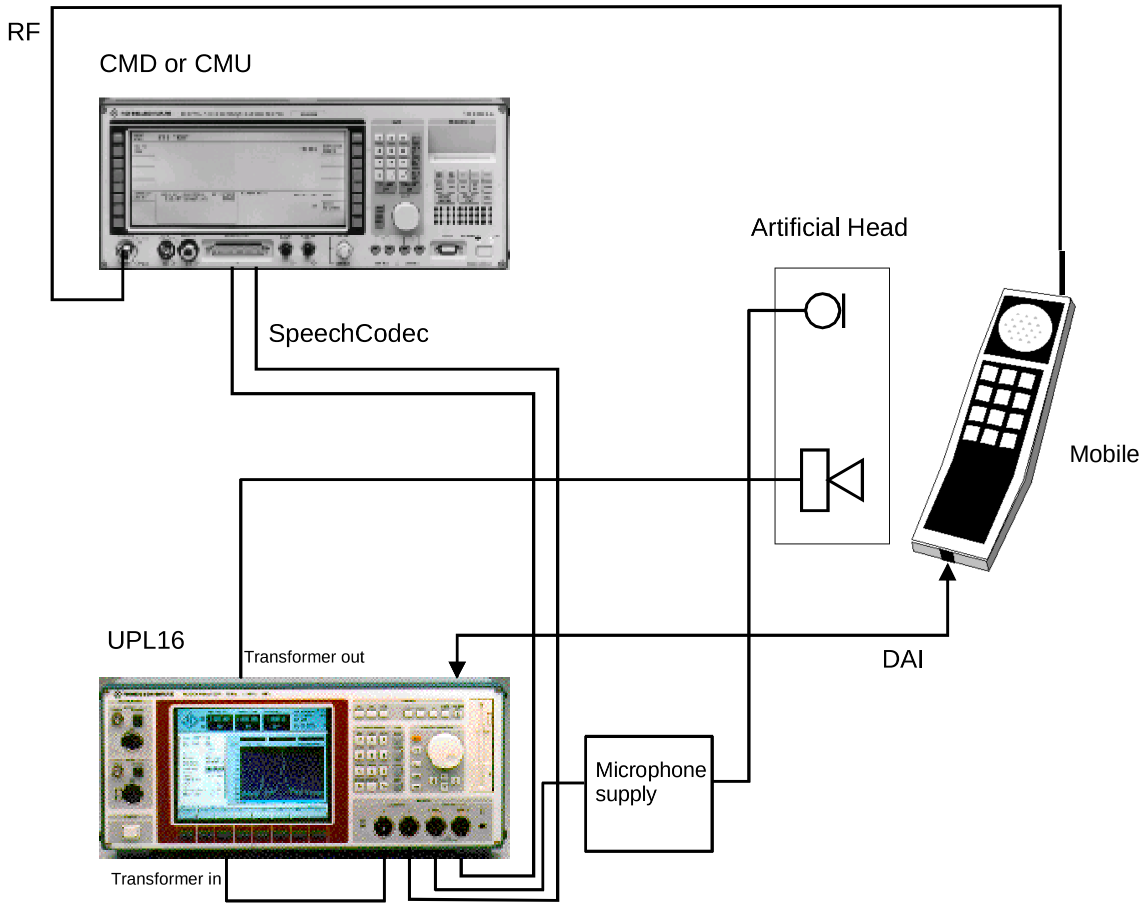


Fig. 4 Test setup UPL with CMD or CMU

3.3 UPL with CMU (validated Testcases Rel-99): see Annex C

These new testcases (available with upgrade UPL-U81 only) are completely described in Annex C as well as installation, test setup, calibration and operation.

3.4 Data Transmission on the DAI

The DAI of the UPL should be configured ad follows (see 4.1.1, UPL used as System Simulator):

Interface	DAI to MS
Reset	START
Test Ctrl	ACOUST DEVS
DAI Source	AUDIO GEN

A call has to be established between the mobile and the system simulator (for signalling).
The DAI of the mobile has to be configured properly to be ready for being switched to the test mode by the system simulator.
Note: The details of these steps depend on the DAI of the mobile to be tested.

If the DAI of the mobile is ready it will be set to the test mode by the UPL:
Set the test control lines: Test Ctrl → ACOUST DEVS
Wait more than 1 second, then send a reset pulse: Reset → START

After the rising edge of the reset pulse the mobile has to send the data clock and at the same time start data transmission. Upon the first clock edge the UPL also starts to transmit data. If the digital generator is selected in the UPL, the signal set in the generator is sent to the mobile and should be audible in the loudspeaker when a suitable level is set.

Upon installation of the UPL software, a test setup DAI_TST1.SAC is stored in the directory C:\UPL\SETUP which allows the function of the DAI interface to be checked in a simple way:

Load setup DAI_TST1.SAC and start sweep:
  The digital generator sweeps the frequency between 300 Hz and 3 kHz.
  The digital analyzer displays the measured signal as FFT.

The tones should be audible in the loudspeaker of the mobile. The echo received by the microphone of the mobile is transmitted to the UPL via the DAI and visible in the spectrum. A whistling tone at the microphone is visible as a line in the displayed spectrum.

3.5 Artificial Ear and Mouth

The artificial ear (microphone) is to be connected to the input of the UPL analyzer via a suitable amplifier or a feed unit.

The artificial mouth (loudspeaker) cannot be fed directly from the output of the UPL generator. It has to be connected to the output via the 4:1 adapter transformer accommodated in the rear panel of UPL16.

3.6 Calibration Procedures

The microphone of the artificial ear has to be calibrated with aid of an sound pressure calibrator producing a sound pressure level of 94 dBspl which equals a sound pressure of 1 Pa at a frequency of 1 kHz eg. B&K 4231 or B&K 4230

3.6.1 Calibration of artificial ear and artificial mouth

3.6.1.1 Using artificial ear ITU-T Recommendation P.57 Type 1

3.6.1.1.1 Ear Calibration

If the used ear type is changed this step must be repeated!

The microphone has to be detached from the parts of the artificial ear (Type 1) (switch power supply of microphone off for disassembly!). After reassembling the measuring microphone and switching on the supply the microphone has to be connected to the sound pressure calibrator which has also to be switched on. After switching on the power supply of the microphone at least 1 minute settling time is needed for the microphone to stabilize The output voltage of the microphone at the output of the microphone preamplifier and power supply is measured and stored as reference for all subsequent measurements.

Note: after calibration the gain of the preamplifier and other settings of the microphone supply and preamplifier must remain unchanged for all measurements!

Calling the routine CAL_MIC measures the microphone output voltage, calculates the internal reference value in dBV/Pa and stores it on a file as reference for all later measurements. The microphone sensitivity in mV/Pa is calculated and shown on the screen for validation with the value delivered with the calibration chart of the microphone.

3.6.1.1.2 Mouth Calibration

The absolute sensitivity and the frequency response of the artificial mouth have to be compensated for all measurements. The calibration of the mouth is done with the measuring microphone of the artificial ear which has to be calibrated first. The microphone must be disassembled from the ear parts.

The microphone has to be mounted in a rectangular position to the mouth reference point using the test fixture parts delivered with the mouth.

Calling the routine CAL_MOU measures the sound pressure level at 1 kHz and adjusts the voltage at the mouth input until the sound pressure equals 1 Pa (94 dBspl). This voltage is stored on file for subsequent measurements as mouth sensitivity value.
In the next step the output voltage of the mouth is set to -4.7 dB relative to the value measured before thus adjusting a sound pressure level of -4.7 dBPa (measuring level for most measurements) and a sweep using the frequencies according to P79 is performed. The measured frequency response is internally stored as an inverted response as equalisation file for the generator. Due to nonlinearities in the mouth speaker small errors may still occur therefore a second response measurement with equalisation switched on is performed and the small residual error factors due to nonlinearities are mesured. The initial equalisation file will be corrected by thmeasured correction factors thus producing a new equalisation.

For validation a third sweep is started using the new equalisation file. The measured sound pressure level is checked for an absolute sound pressure of -4.7 dBPa ±0.2 dB. The display shows the response and the limit lines together with a PASS or FAIL indication.

3.6.1.2 Using artificial ear ITU-T Recommendation P.57 Type 3.2

3.6.1.2.1 Ear Calibration Step 1

If the used ear type is changed this step must be repeated!

The microphone in the artificial ear has to be calibrated with aid of an sound pressure calibrator producing a sound pressure level of 94 dBspl which equals a sound pressure of 1 Pa at a frequency of 1 kHz eg. B&K 4231 or B&K 4230

The microphone of type 3.2 artificial ear cannot be detached! The ear has to be calibrated using adapter DP 0939 for the calibrator. After switching on the power supply of the microphone at least 1 minute settling time is needed for the microphone to stabilize. The calibrator has to be switched on and the adaptor DP0939 has to be attached to the pinna of the artificial ear in a sealed condition. The output voltage of the microphone at the output of the microphone preamplifier and power supply is measured and stored as reference for all subsequent measurements.

Note: ***after calibration the gain of the preamplifier and other settings of the microphone supply and preamplifier must remain unchanged for all measurements!***

Calling the routine C32L_MIC measures the microphone output voltage, corrects the sound pressure reference due to calibration value of adapter DP0939, calculates the internal reference value expressed in dBV/Pa and stores it on a file as reference for all later measurements. The microphone sensitivity in mV/Pa is calculated and shown on the screen for validation with the value delivered with the calibration chart of the microphone.

3.6.1.2.2 Ear Calibration Step 2

The individual calibration data of the artificial ear are needed for all subsequent measurements, those data are delivered on a floppy disc together with the artificial ear. The artificial ear according to ITU-T Recommendation P.57 type 3.2 simulates a human ear with the microphone placed at the DRP (Drum Reference Point), for measurements according to 3GPP TS 51.010 chapter 30 the sound pressure level has to be measured at the ERP (Ear Reference Point) so each measured value has to be transformed from DRP to ERP using the calibration data of the ear.

Calling the routine C32L_EAR reads the data from the calibration disc and stores the correction values in the internal file EAR_32L.CAL on the harddisc of UPL16. The routine automatically checks the file name OES_LL.ADA and data format on the calibration disc and displays PASS if the values are correct.

The routine does not check the date of the calibration disc and the calibration values itself, so the user is responsible to insert the calibration disc with actual valid calibration data!

The calibration data are stored on the harddisc of the UPL16. It is not necessary to repeat this step as long as the calibration data of the artificial ear type 3.2 remain unchanged!

3.6.1.2.3 Mouth Calibration

The calibration files for the artificial mouth are identical for both ear types. So if the mouth was already calibrated it is not necessary to recalibrate it if the used ear type is changed!

The absolute sensitivity and the frequency response of the artificial mouth have to be compensated for all measurements. For the calibration of the mouth an additional measuring microphone is needed which has to be calibrated first. The preamplifier used with the artificial ear can be used but an additional microphone capsule type 4134 is needed (the microphone capsule of artificial ear type 1 can be used if available).

- Switch power supply of microphone off and unscrew microphone preamplifier from artificial ear type 3.2.
- Complete measuring microphone with additional microphone capsule to form a usual measuring microphone and switch power supply of microphone on. Allow at least 1 minute setup time for the microphone.
- Attach this microphone to the sound level calibrator, switch calibrator on and call routine C32L_MMO.
- Mount this microphone in a rectangular position to the mouth reference point using the test fixture parts delivered with the mouth.

Calling the routine C32L_MOU measures the sound pressure level at 1 kHz and adjusts the voltage at the mouth input until the sound pressure equals 1 Pa (94 dBspl). This voltage is stored on file for subsequent measurements as mouth sensitivity value.
In the next step the output voltage of the mouth is set to -4.7 dB relative to the value measured before thus adjusting a sound pressure level of -4.7 dBPa (measuring level for most measurements) and a sweep using the frequencies according to P79 is performed. The measured frequency response is internally stored as an inverted response as equalisation file for the generator. Due to nonlinearities in the mouth speaker small errors may still occur therefore a second response measurement with equalisation switched on is performed and the small residual error factors due to nonlinearities are mesured. The initial equalisation file will be corrected by the measured correction factors thus producing a new equalisation.

For validation a third sweep is started using the new equalisation file. The measured sound pressure level is checked for an absolute sound pressure of -4.7 dBPa ±0.2 dB. The display shows the response and the limit lines together with a PASS or FAIL indication.

The frequencies for calibrating the artificial mouth are stored in the CAL.SPF file.

101	212	450	950	2002	4250
106	224	475	1002	2120	4500
112	236	502	1060	2240	4750
118	251	530	1120	2360	5000
126	265	560	1180	2500	5300
132	280	600	1250	2650	5600
140	300	630	1320	2800	6000
150	315	670	1400	3000	6300
161	335	710	1500	3150	6700
170	355	750	1602	3350	7100
180	375	802	1700	3550	7500
190	402	850	1800	3750	8000
201	425	900	1900	3950	

3.6.2 Calibration of the Speech Codec in the CMD or CRTx

Some measurements have to be performed via the RF interface and thus via the speech coder/decoder. The gain of the speech path should also be calibrated since the gain is entered directly into the measurement result. For this calibration it is necessary to close the RF path via a mobile and to set the mobile to a so-called loop-C status. The audio data sent to the mobile are directly returned via the RF interface. This nearly corresponds to a total echo or an echo path with a gain of 0 dB.

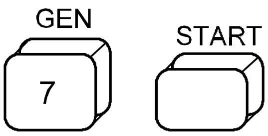
The CODEC-CAL routine sets this status or requests the user to perform this setting and then carries out an echo loss measurement in compliance with test case 30.6.1 with artificial speech. The measured value directly corresponds to the gain of the speech codec and is stored in the CODEC.CAL file.

4 Manual Operation

Note: *No special knowledge of the MS-DOS operating system is required for use of the UPL.*

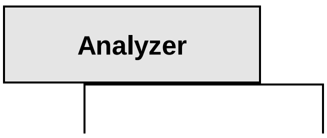
It is assumed that you know what is meant by eg a file, a directory or a path and no further explanations are provided on that.

Legend of graphic symbols used in this manual:

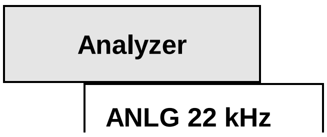


Front-panel keys

Softkey



Menu item of a panel



Parameter of a menu item

4.1 Configuration of the DAI

The operating status of the GSM digital audio interface (DAI) is set under

DIGITAL I/O CONFIG in the OPTIONS panel.

Interface

Selects the operating mode of the interface.

DAI to MS

The UPL acts as a system simulator and can be connected to a mobile. This operating mode, defined in 3GPP TS 44.014 section 10, should always be used when connecting the UPL to a DAI-Mobile to perform the acoustic test cases.
The UPL *sends* a reset pulse and then *waits* for the data clock to start data transmission.

DAI to SS

UPL simulates a mobile (MS) and can be connected to a system simulator (SS). The direction of the signals on the DAI is inverted as against the DAI to MS status: the UPL *waits* for a reset pulse and then *sends* the data clock at the beginning of data transmission.
In this mode a digital link can be set up to instruments the DAI of which is permanently configured as system simulator.
This allows the UPL to be connected for instance to the digital interface of the speech coded of a system simulator.
The test control lines are switched to high impedance.

AUX to SS

The second (auxiliary) audio interface is used. It has the same function as the DAI to SS interface but another pin assignment.
Only the pins designated "not used" in 3GPP TS 44.014 are used for this additional interface.

4.1.1 UPL used as System Simulator (DAI to MS)

Interface

DAI to MS

Reset

Activates the operating status of the mobile after setting the test control lines and starts data transmission to the interface.

START

Triggers a reset pulse: the status of the reset line changes from high to low and to high again after approx. 5 ms.

Test Ctrl

The two test control lines determine the operating mode of the mobile. Having set the appropriate test control signals, a reset pulse has to be applied to activate the test mode in the mobile.
This function is only available in the "Interface DAI to MS" mode. If DAI to SS or AUX to SS is selected, the lines are at high impedance.

NORMAL MODE

The mobile is in normal operating mode (idle mode)
Test control line 1 = Low
Test control line 2 = Low

SPEECH DEC

Test of speech decoder / DTX functions (downlink)
Test control line 1 = Low
Test control line 2 = High

SPEECH ENC

Test of speech encoder / DTX functions (uplink)
Test control line 1 = High
Test control line 2 = Low

ACOUST DEVS

Test of acoustic devices and A/D & D/A
Test control line 1 = High
Test control line 2 = High

DAI Source

Selects the signal source for audio data from the UPL to the mobile.
This function is only available in the DAI to MS interface mode. With DAI to SS or AUX to SS selected, the digital generator of the UPL is always the source for the audio data.

AUDIO GEN

When the digital instrument is selected in the GENERATOR panel, the UPL sends the signal set in the panel on the DAI. The signal 0 is output when the analog instrument is selected.

PCM Value 1

The signal is the PCM value 1 specified for certain test cases, irrespective of the generator setting. The LSB is set at each second data word, all other bits are 0.

4.1.2 UPL used as Mobile Simulator

Interface

DAI to SS
AUX to SS

The following items can be selected:

Mobile

Selects the behaviour of UPL as mobile on the DAI.
Two operating modes can be selected: mobile NORMAL or SPECIAL.

NORMAL

The UPL acts as defined by 3GPP TS 44.014 for a mobile on the DAI: UPL *waits* for a reset pulse and then *sends* the data clock at the beginning of data transmission. Since this mode is permanently defined, no other settings can be made under NORMAL.
Note: the characteristics of the NORMAL mode can also be set in the SPECIAL mode.

SPECIAL

Should the UPL be connected to a DAI which does not exactly corresponds to 3GPP TS 44.014 specifications, the characteristics of the reset and clock signals can be changed here.

Reset

.

START

UPL *sends* the reset pulse upon confirmation of START.

INPUT

UPL *waits* for the reset pulse same as in the NORMAL mode.

Test Ctrl

This function is only available in the "Interface to DAI to MS" mode. With DAI to SS or AUX to SS selected, the lines are at high impedance.

Clock

.

INPUT

UPL *waits* for the data clock.

INPUT INV

UPL *waits* for the *inverting* data clock: the data are taken over with the falling edge and change at the rising edge.

OUTPUT

UPL *sends* the data clock same as in the NORMAL mode.

OUTPUT INV

UPL *sends* the *inverting* data clock: the data are taken over with the falling edge and change at the rising edge.

DAI Source

This function is only available in the "Interface DAI to MS" mode. With DAI to SS or AUX to SS selected, the digital generator of the UPL is always the signal source for the audio data.

4.2 Generators (GENERATOR Panel)”)

Activating the GENERATOR panel

- *UPL front panel*: GEN
- *External keyboard*: ALT + G
- *Mouse*: (repeated) clicking on the panel name, until the generator panel is displayed.

If the GENERATOR panel is already displayed on the screen, it can also be activated by actuating one of the TAB keys (repeatedly) or by a mouse click.
Advantage: The panel need not be set up again.

The GENERATOR panel is always displayed on the left side of the screen and consists of two sections: configuration and function.

GENERATOR		
GENERATOR	DIGITAL	Select instrument (analog or digital).
CHANNEL(s) : : : :	CH1	Configuration section for setting the outputs (output connectors, channel select, output impedance, etc.) see UPL Manual, section 2.5.2 Configuration of the Analog Generator see 4.2.2 Configuration of Digital Generator.
FUNCTION :	SINE	Functions (waveforms) of the generator, see UPL Manual, section 2.5.4

On changing the *function* (generator signal)

- the current function is stored on the hard disk;
- the desired function is loaded from the hard disk, initialized and, if possible, started.

On changing the *generator* (eg **DIGITAL** instead of **ANLG**)

- the current generator with all settings and the currently active function is stored to the hard disk;
- the desired generator with the currently active function is loaded from the hard disk, initialized and, if possible, started.

***Note**: The "parameter link" function, which can be selected in the OPTION panel, may be used to influence the UPL upon changes of function and instrument. If desired, settings in the function and/or configuration section of the generator panel are used for the new function or instrument - if physically possible. A change of instrument from ANLG to DIGITAL can be performed, for instance, without the function and its frequency parameters changing in the panel.*

4.2.1 Selecting the Generator

GENERATOR

The GENERATOR panel contains the settings for the analog and the digital generators.

ANLG

Two-channel analog generator, 2 Hz to 21.75 kHz, with low-distortion generator (UPL-B1 option) up to 110 kHz

DIGITAL

Single-channel digital generator, 2 Hz to 3.96 kHz

The standard analog generator can be supplemented by the low-distortion generator option (UPL-B1) for sinewave generation in the analog range, thus allowing for generating a particular low-distortion sinewave signal with a frequency of up to 110 kHz.

In the digital mode this low-distortion generator can also be used as an analog auxiliary generator so that digital and analog sinewave signals can be simultaneously generated with option UPL-B1.

Frequency range of the digital generator:

The generator attains 99% of the Nyquist frequency. The sample frequency is 8 kHz and displayed in the configuration section of the GENERATOR panel using the menu item Sample-Frq. So the maximum generator output frequency is given by:

$$\boxed{f_{max} = 0.5 \text{ x sample frequency x } 0.99 = 3.96 \text{ kHz}}$$

There are 3 states the active generator (visible in the panel) can assume (see UPL Manual, section 2.3.5 Status Display):

- RUNNING: The set function (generator signal) is output via the specified interface.
- BUSY: The generator output signal is calculated.
- HALTED: The generator is halted, no output signal; the outputs are terminated.

RUNNING is the normal status of the generator, ie a signal is constantly output. The generator is automatically restarted after a reset.

The generator can be manually restarted at all times by calling the generator or the function (open the respective selection window and confirm using ENTER). This may be required with burst signals to immediately start a new interval (with the burst phase).

The setting of some signals (eg specific noise signals) involves a lot of computations. During this time, the generator does not produce any signal and is in the BUSY state. After having successfully concluded the computations, the generator automatically re-enters the RUNNING state. If further settings are made or keys are pressed during computation, the computation is aborted and automatically restarted. The generator shortly assumes the HALTED state.

Other reasons for halting the generator (HALTED):

- Faulty setting (eg wrong file name for an equalization or sweep list).
  Remedy: eliminate the cause of error; restart, if required.
- On the digital instrument:
  Applying a too high external frequency (generator is ”overrun”).
  Remedy: reduce the external clock frequency and restart.

4.2.2 Configuration of the Digital Generator

Coding

The generator signal can be subjected to a coding simulation which routes the signal through a coder - decoder path.

PCM LINEAR No coding simulation; the signal is output with a 13-bit PCM linear coding.

A-LAW SIMUL The generator signals is A-law coded and decoded again.

Channel(s)

Only 1 channel is available which can be switched on and off.

OFF Generator switched off

1 Generator switched on

Sample Frq

Display of the nominal output clock rate.

8 kHz The digital generator can only be operated with a (fixed) clock rate of 8 kHz. The clock may be internally generated or externally applied. The signal source is the same for digital generator and analyzer and can be selected in the OPTIONS panel.

Note: The applied digital (generator/analyzer) clock is continually measured in the UPL irrespective whether one of the two digital instruments is active or not. The measured clock frequency can be read out as SENS4:DATA via the automatic sequence controller or by remote control. For a display in the manual mode, the analyzer has to be set to DIGITAL and the frequency measurement to SAMPLE FREQ.
If no clock is present, an NAN value (not a number; defined in the UPL as $9.93*10^{37}$) is supplied.

Audio Bits

Display of word width of generated audio samples in bits.

The GSM DAI works on a fix word width of 13 Bits

4.2.3 Functions

Function	
SINE	Single sine, dither may be included
MULTISINE	Up to 17 sines
SINE BURST	Sine burst signal
SINE^2 BURST	Asymmetrical sine burst
MOD DIST	Test signal for intermodulation distortions
DFD	Test signal for difference frequency distortions
RANDOM	Random noise
ARBITRARY	Arbitrary waveform
POLARITY	Test signal for polarity measurements
FSK	Frequency shift keying, only with UPL-B33 installed; only required for ITU-T O33 (via automatic sequence controller).
RANDOM+ANLR	Random noise with feedback of analyzer signal; required for acoustic GSM measurement "stability margin".

4.2.3.1 Special Features of Digital Generator Functions

The main differences of model UPL 16 as against UPL 06 with option UPL-B2 are caused by the lower sampling rate and the narrower word width.

The lower sampling rate of 8 kHz reduces the available frequency range to 3.96 kHz but it improves the frequency resolution for the noise signals (RANDOM Domain FREQ, RANDOM+ANLR, MULTI SINE) by the same degree (below 0.5 Hz). The limit frequencies for each frequency menu item are displayed in the user information line so that further explanations are not required. Since the word width is limited to 13 bits (audio bits), the level resolution is reduced to -72.25 dBFS.

4.2.3.2 Artificial Voice

The artificial voice (male and female) is used whenever an excitation signal is required which comes as near as possible to reality. Such signal is required for echo loss measurements with the acoustic GSM test cases.

The artificial voice is created via the generator function ARBITRARY. Two special waveform files (one for male the other for female) in CPR format can be loaded and reproduced. They have the designations

- P50M.CPR for **m**ale and
- P50F.CPR for **f**emale

The CPR files are sent to the generator DSP online and in packed form. The wider the word width of the samples, the more data have to be downloaded. For this reason the word width of the artificial voice files is reduced to the 13 bits required for GSM. Since the artificial voice signal is an analog signal, the sampling rate must be 48 kHz.

For further information on the generator function used and the CPR format see UPL Manual, section 2.5.4.10 ARBITRARY (User-programmable Signal).

4.2.3.3 Feedback of Analyzer Signal with Additional Noise

With the RANDOM+ANLR generator function, the feedback analyzer signal whose gain is selectable is superposed on a noise signal whose frequency can be defined. This function is required for the stability-margin measurements in the acoustic test cases.

The noise signal is defined by specifying an amplitude-frequency distribution in the frequency domain (FREQ). This noise is always white.

The superposed analyzer signal can be taken from measurement channel 1 or 2.

This special signal can only be generated if the "Volt Range" in the GENERATOR panel is set to FIX. Otherwise the loop signal is set to zero gain (muting), while the noise signal remains active.

DC Offset

See UPL Manual, section 2.5.4.1.2 Common Parameters for All Generator Functions

Spacing

Definition of the frequency grid, ie the spacing between the frequency lines.

USER DEF

Manual setting of frequency grid. Any value entered is adjusted to match the nearest settable value. The limits and the settable frequency values depend on the sampling rate (see UPL Manual, section 2.5.1 Selecting the Generator) and the generator selected. The lower frequency limit for the digital instrument is the ratio

system clock / 16384

Units: Hz, kHz

ANLR TRACK

The analysis grid value of the FFT is transferred automatically. This value is also displayed on the ANALYZER panel under "FFT:Resolution" (see UPL Manual, section 2.6.5.12 FFT). This is the optimal setting for an analysis with the rectangular window. If FFT is not selected in the analyzer, the setting is rejected (error message).

Crest Fact

Selecting the algorithm to define the phase of each of the frequency lines and so the crest factor of the total signal.

OPTIMIZED

Automatic *minimization of the crest factor* by internally optimizing each of the phases.

VALUE:

Entering the *desired crest factor.* The phases of the frequency lines are modified internally such that the resulting crest factor closely approximates the desired value. The accuracy of this method depends on the total number of lines thus spacing and frequency range

Lower Freq

Setting the lower frequency limit for the generated noise.

Range: 350 Hz to 550 Hz - 1 x spacing

Upper Freq

Setting the upper frequency limit for the generated noise.

Range: lower freq + 1 x spacing to 550 Hz

RND PEAK

Setting the peak output level of the noise signal. The value input here will *not* affect the loop signal.

VOLT PEAK and VOLT RMS are coupled via the crest factor (which is constant for a particular noise signal). Therefore, if VOLT PEAK is modified, the value of VOLT RMS will also change immediately. If the crest factor is changed, RND PEAK will remain unchanged.

Specified range: 0 to V_{max}
digital: V_{max} = 1 FS
analog: V_{max} = 14,142 V for UNBAL
V_{max} = 28,284 V for BAL

Units:
digital: FS | %FS | dBFS | LSBs | Bits | Δ% | dBr
analog: V | mV | μV | V/V_r | dBu | dBV | dBr | dBm |
Δ%V | ΔV | ΔmV | ΔμV

RND RMS

Setting the RMS output voltage (for analog generator only). The value input here will *not* affect the loop signal.

VOLT PEAK and VOLT RMS are coupled via the crest factor (which is constant for a particular noise signal). Therefore, if VOLT RMS is modified, the value of VOLT PEAK will also change immediately.

Units:
digital: FS | %FS | dBFS | LSBs | Bits | Δ% | dBr
analog: V | mV | μV | V/V_r | dBu | dBV | dBr | dBm |
Δ%V | ΔV | ΔmV | ΔμV

***Note**:* *VOLT RMS can only be entered while the generator is RUNNING. When the noise signal is being calculated (GEN BUSY), the crest factor is still not known; entries for VOLT RMS will therefore be rejected at this stage and the voltage registered as 0.0. To make sure the noise signal calculated by the generator is output at the correct (peak) amplitude, it is best to enter a value under VOLT PEAK, which can be done at any time.*

Loop Chan

Specifying the analyzer channel whose signal is to be fed back to the generator and superposed onto the noise signal.

OFF	Feedback is switched off. Feedback amplification can be selected as a presetting even if loop is switched off.
1	Channel 1 is fed back.
2	Channel 2 is fed back.

Loop Gain

Setting the feedback gain

Range: 0 to 1000

Units: * (dimensionless factor) | dB

***Note**: As soon as "Volt Range" (on the GENERATOR panel) is switched to AUTO, loop gain is automatically set to 0 by the program (muting).*

4.2.4 Auxiliary Generator

With option UPL-B1 (low-distortion generator) fitted, an additional sinewave generator is available for producing analog signals up to 110 kHz. Thus, *digital* audio data of any signal shape and an *analog* sinewave signal can be generated *simultaneously.*

The auxiliary generator has the same specifications as the low-distortion generator and its own (1-dimensional) sweep system for sweeping either the level or frequency. The generator can be used as a balanced or unbalanced source with one or two output channels. Different source impedances can be selected. Level control is via the output amplifier.

AUX GEN

Activation of auxiliary generator.

OFF	Auxiliary generator switched off; only digital audio data are generated, analog outputs switched off.
ANALOG OUT	In addition to the digital signal an analog signal is generated at the analog XLR connector. Frequency and level of the analog signal can be set or swept.

All other functions of the auxiliary generator when used as analog generator are unchanged and can be looked up in the UPL manual (see UPL Manual, section 2.5.5.1 Auxiliary Generator used as Analog Generator.

4.3 Analyzers (ANALYZER Panel)

Activate the ANALYZER panel:

UPL front panel:	ANLR key
External keyboard:	ALT + A
Mouse:	(repeated) click on the panel name, until the ANALYZER panel is displayed

If the ANALYZER panel is already visible on the screen, it can also be activated by actuating one of the TAB keys (repeatedly) or by a mouse click.
Advantage: The panel need not be set up again.

4.3.1 Selecting the Analyzer

ANALYZER	The ANALYZER panel provides the settings for 2 analog and 1 digital analyzer instrument.
ANLG 22 kHz	2-channel analog analyzer, DC/10 Hz to 21.9 kHz
ANLG 110 kHz	2-channel analog analyzer, DC/20 Hz to 110 kHz
DIGITAL	1-channel digital analyzer, DC/10 Hz to 3.656 kHz

The ANALYZER panel consists of the following sections:

ANALYZER			
ANALYZER	ANLG 22k Hz	Configurations	Selection of the analog or digital instrument, reference impedance for power units, configuration section for setting the test inputs.
CHANNEL(s)	2 ≡ 1	Configurations	(Input connectors, channel selection, input impedance) see 4.3.2 Configuration of Digital Analyzer see UPL Manual, section 2.6.2 Configuration of the Analog Analyzers
START COND	AUTO	Configurations	Ways of starting the analyzer, see UPL Manual, section 2.6.4
INPUT DISP	ON	Higher-level functions	Input signal, see UPL Manual, section 2.6.5.18 INPUT
FREQ/PHASE	FREQ & PHASE	Higher-level functions	Combined frequency / phase measurement, see UPL Manual, section 2.6.5.19
FUNCTION	RMS & S/N	Functions	Analyzer functions, see UPL Manual, sections 2.6.5.2 to 2.6.5.19

When switching from the analyzer instrument to the other, the data of all segments are stored for the current instrument, the data of the new instrument are loaded and the panel contents can be entered anew. When changing to the analyzer function, the settings in the configuration range are retained.

Note: *The "parameter link" function which can be selected in the OPTION panel may be used to influence the UPL upon changes of function and instrument. As requested, existing settings in the function and/or configuration section of the GENERATOR panel are accepted for the new function or instrument - if physically possible.*

Table 4: Measurement range limits of the ANALYZER instruments:

Instrument	**Lower limit**	**Upper limit**	**Sample rate**
ANLG 22 kHz[1)]	DC/10 Hz	21.9 kHz	48 kHz
ANLG 110 kHz[1)]	DC/20 Hz	110 kHz	307.2 kHz
DIGITAL	DC 10 Hz	3.65 kHz	8 kHz

[1)] The frequency value refers to the upper limit of the analog analyzers

Lower limit:

DC: Setting the DC function in one of the two analog analyzer instruments results in DC coupling of the input unit

10 Hz: The menu item "Min Freq" in the analyzer instruments ANLG 22 kHz and DIGITAL indicates the lower limit.

20 Hz: The menu item "Min Freq" in the "fast" analyzer instruments (ANLG 110 kHz) indicates the lower limit.

Upper limit:

Signals can be measured up to this limit.

Measurement range limits of the digital ANALYZER instrument:

The maximum measurement frequency is given by

f_{max} *= sample frequency x 0.5 for RMS, otherwise sample frequency x 117 / 256*

The sample frequency is fixed to 8 kHz.

Table 5: Availability of functions depending on the ANALYZER instrument:

Instrument	Measurement functions																
	RMS	RMSsel	PEAK	QPEAK	DC	THD	THD +N	MOD DIST	DFD	Wow &FL	FFT	Coher	Rub & Buzz	Polarity	Filter simul.	1/3-Octave	WAVE-FORM
ANLG 22 kHz	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes
ANLG 110 kHz	yes	yes	-	-	yes	yes	yes	yes	yes	-	yes	-	yes	yes	yes	-	yes
DIG ITAL	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	yes	-	-	yes	yes	yes	yes

Higher-level functions:

The selected function can be complemented by higher-level supplementary functions.

INPUT-DISP measurement:

see UPL Manual, section 2.6.5.18

- PEAK display of peak value of the two input signals
- RMS display of rms value for the measurement functions THD, THD+N, FFT, MOD DIST and DFD

If INPUT DISP RMS is set and a measurement function is selected which does not allow for RMS display or where it does not make sense, "-----" is displayed in the Input RMS window. INPUT PEAK measurements can still be performed.
The availability of the INPUT RMS measurement related to the selected measurement function can be looked up in the subsequent table.

Frequency and phase measurement

see UPL Manual, section 2.6.5.19

- FREQuency display on all switched-on channels

Additionally, in analyzer ANLG 22 kHz

- FREQuency display on channel 1, PHASE display on channel 2; selectable only with two-channel measurement
- FREQuency display on channel 1, GRPDEL group delay) display on channel 2, selectable only with two-channel measurement

***Note**: Since the digital analyzer is a 1-channel analyzer, phase or group delay measurements cannot be performed with it.*

If a measurement function has been selected which does not allow frequency or phase measurement or where it does not make sense (such as DFD), "-----" is displayed in the "Freq & Phase“ window. Refer to the table below for the availability of the frequency and phase measurement in dependence of the selected measurement function.

Table 6: Availability of frequency and phase measurement depending on measurement function

Measurement function	**INPUT DISP**		**FREQ/PHASE**		
	PEAK	**RMS**	**FREQ**	**PHASE**	**GRP DELAY**
OFF	yes	yes	yes	ANLG 22kHz	ANLG 22kHz
RMS & S/N	yes	ANLG 110 kHz	yes	ANLG 22kHz	ANLG 22kHz
RMS select	yes	yes	yes	no	no
PEAK & S/N	yes	no	no	no	no
Q PK & S/N	yes	no	no	no	no
DC	yes	no	no	no	no
THD	yes	yes	yes	no	no
THD+N	yes	yes	yes	ANLG 22kHz	ANLG 22kHz
MOD DIST	yes	yes	no [1)]	no	no
DFD	yes	yes	no[1)]	no	no
WOW & FL	yes	no	no	no	no
POLARITY	yes	no	no	no	no
FFT	yes	yes	yes	ANLG 22kHz	ANLG 22kHz
FILTSIM	no	no	no	no	no
WAVEFORM	yes	no	no	no	no
Coherence	yes	no	no	no	no
1/3-OCTAVE	yes	no	no	no	no
Rub & Buzz	yes	ANLG 110 kHz	yes	no	no

1) However, the individual frequencies of signal and interference are displayed in a histogram or spectrum list.

4.3.2 Configuration of Digital Analyzer

***Notes**:*

- *Prior to making analyzer settings, the generator should be configured usefully. Otherwise, undesired generator settings could cause superfluous restrictions in the analyzer panel.*
- *If the generator is not used as a signal source, it is advisable to switch over to analog mode in order to avoid any interference of the generator settings.*

Coding

The received signal can be subjected to a coding simulation before it is handled by the measurement routines.

PCM LINEAR No coding simulation; the signal is processed unchanged.

A-LAW SIMUL The signal is routed through an A-law coder - decoder path before it is measured by the analyzer.

Min Freq

10 Hz

Display of the lower frequency range limit for measurement of the digital audio data.

Sample Frq

8 kHz

Display of (nominal) analyzer clock rate .

The digital analyzer can only be operated with a (fixed) clock rate of 8 kHz. The clock may be internally generated or externally applied. The signal source is *the same* for digital generator and analyzer and can be selected in the OPTIONS panel,.

***Note**:* *The applied digital (generator/analyzer) clock is continually measured in the UPL irrespective whether one of the two digital instruments is active or not. The measured clock frequency can be read out as SENS4:DATA via the automatic sequence controller or by remote control. For a display in the manual mode, the analyzer has to be set to SAMPLE FREQ.*
If not clock is present, a NAN value is supplied.

The maximum measurement frequency of the digital analyzer is obtained from the sample frequency and the (modified) "Nyquist" factor:

$$f_{max} = \text{sample rate} \times 117 / 256 = 3{,}65625 \text{ kHz}$$

Audio Bits

Display of word width of received audio samples in bits.

The GSM DAI works on a fix word width of 13 Bits

4.3.3 Special Features of Digital Analyzer Functions

The main differences of model UPL 16 as against UPL 06 with option UPL-B2 are due to the lower sampling rate and the narrower word width.

The lower sampling rate of 8 kHz reduces the available frequency range to 3.66 kHz but it improves the frequency resolution of the FFT-supported measurement functions (THD+N, DFD, MOD DIST, FFT) to the same degree (below 0,5 Hz in the whole frequency range; 7.6 mHz at max. zoom depth). The limits for frequency entries under each frequency menu item are displayed in the user info line so that no further explanations are required.

Since the word width is limited to 13 bits (audio bits), the level resolution is reduced to -72.25 dBFS.

The measurement function COHERENCE is a 2-channel measurement it cannot be selected.

4.4 Analyzer Filters

The FILTER panel has been devised for definition of the filters which can then be used in the ANALYZER panel. Before you can select a user-definable filter in the analyzer, you must, of course, create it in the FILTER panel.

Activating the FILTER panel:
UPL front panel: FILTER
External keyboard: ALT + T
Mouse: (repeated) click on the right panel name, until the FILTER panel is displayed.

If the FILTER panel is already displayed on the screen, it can be activated also by actuating one of the TAB keys (repeatedly) or by a mouse click.
Advantage: The panel need not be set up again.

Filter

For the functions

- RMS & S/N (rms measurement) → 3 filters possible
- PEAK & S/N (peak measurement), → 3 filters possible
- Q-PK & S/N (quasi-peak meas.) → 3 filters possible
- THD+N/SINAD (distortion measurement) → 1 filter possible
- RMS SEL (selective RMS measurement) → 1 filter possible
- FILTSIM (filter simulation) → 3 filters possible
- RUB & BUZZ (loudspeaker measurement) → 1 filter possible
- WAVEFORM (Time domain display) → 1 filter possible
- 1/3-OCTAVE → 1 filter possible

Any desired filter from the filter selection window can be set in the ANALYZER panel. This window contains user-definable filters (the first 9) and weighting filters, which are referred to by their short names in the FILTER panel or by a name complying to the standard. You can select any desired filter (also several times) and assign it to the ANALYZER measurement function.

The sum frequency response of all selected filters can be graphically displayed using the FILTSIM analyzer function (see UPL Manual, section 2.6.5.13).

***Note**: In the WAVEFORM mode, no filter can be selected in the 110 kHz analyzer.*

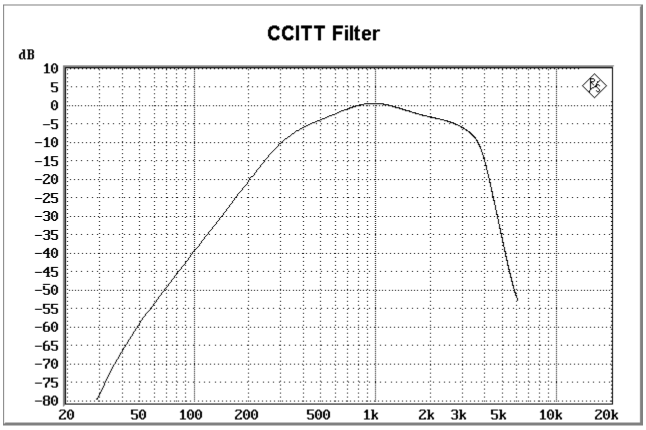
4.4.1 Weighting Filters

The user-definable filters in the UPL (see UPL Manual, section 2.7.2 Creating the User-definable Filters) are complemented by a set of pre-defined weighting filters, which are automatically matched to the current sample rate. In the analog analyzer 15 different filters can be selected; in the digital analyzer only the CCITT filter because of the low sampling rate.

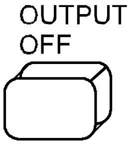
Note: *The weighting filters are always available with the filter command of the respective measuring function and need not be set in the FILTER panel.*

Filter: CCITT

Standard(s): CCITT 0.41
IEEE Rec. 743-84
CISPR 6-76
CCITT Rec. P.53



4.5 Fast Switch-off of Outputs



Switches off all outputs.

- Digital output set to OFF (0 FS).
- Analog outputs are terminated (impedance is retained);
  output level = 0 V.

An LED indicates the state of the key. LED on signifies OUTPUT OFF. The generator status is also indicated as "Output Off" (yellow) in the status line so the state is known even if an external monitor is used and no attention is paid to the UPL.

The outputs are *automatically switched off* when

- the generator is overloaded or
- the analyzer outputs are overloaded (input voltage into 300 Ω and 600 Ω > 25 V); in this case the input impedance is internally switched to 200 kΩ.

When switched off, the lines can be reactivated only by pressing the OUTPUT OFF key again. After an overload of analyzer inputs, measurements can be continued if the overload has been eliminated and the input impedance switched to 200 kΩ.

5 Remote Control

5.1 Configuration of DAI (Digital Audio Interface)

Command	Parameter	Basic unit	Meaning	Section
CONFigure:***DAI***:***INT****erface*	***DAIM****s*		→ UPL acts as a system simulator (SS) and is connected to a mobile (MS). The operating mode, defined in 3GPP TS 44.014 section 10, should be used for the acoustic test cases. UPL *sends* a reset pulse and then *waits* for the data clock to start data transmission.	**4.1** **OPTIONS panel** Interface → DAI to MS → DAI to SS → AUX to SS
	DAIS*s*		→ UPL simulates a mobile (MS) and can be connected to a system simulator (SS). The direction of signals on the DAI is reversed as against the DAI to MS status: UPL *waits* for a reset pulse and *sends* the data clock at the beginning of data transmission.	
	AUXS*s*		→ The second (auxiliary) audio interface is used. It has the same function as the DAI to SS interface but a different pin assignment. Pins designated "not used" in 3GPP TS 44.014 are used for this additional interface.	

5.2 UPL used as System Simulator (DAI to MS)

Command	Parameter	Basic unit	Meaning	Section
CONFigure:***DAI***:***INT****erface*	***DAIM****s*		→ UPL acts as a system simulator (SS) and is connected to a mobile (MS). The operating mode, defined in 3GPP TS 44.014 section 10, should be used for the acoustic test cases. UPL *sends* a reset pulse and then *waits* for the data clock to start data transmission	**4.1.1** **OPTIONS panel** Interface → DAI to MS
CONFigure:***DAI***:***RES****et*	***GO***		Activates the operating status of the mobile after setting the test control lines and starts data transmission to the interface. → Triggers a reset pulse: the status of the reset line changes from high to low and to high again after approx. 5 ms.	**4.1.1** **OPTIONS panel** Reset → START
CONFigure:***DAI***	***TST0*** ***TST1*** ***TST2*** ***TST3***		Test control line 1 and test control line 2 determine the operating mode of the mobile. After setting the appropriate test control signals, a reset pulse has to be applied to activate the test mode in the mobile. → The mobile is in normal operating mode (idle mode). Test control line 1 = Low Test control line 2 = Low → Test mode for measurements of speech decoder (downlink) Test control line 1 = Low Test control line 2 = High → Test mode for measurements of speech encoder (uplink) Test control line 1 = High Test control line 2 = Low → Test mode for acoustic measurements Test control line 1 = High Test control line 2 = High Available only in operating mode "Interface DAI to MS" (CONF:DAI:INT DAIM). In operating mode "Interface DAI to SS or AUX to SS (CONF:DAI:INT DAIS|AUXS) the lines are switched to high impedance.	**4.1.1** **OPTIONS panel** Test Ctrl → NORMAL MODE → SPEECH DEC → SPEECH ENC → ACOUST DEVS

Command	Parameter	Basic unit	Meaning	Section
CONFigure:***DAI***:***SOUR****ce*			Selects the signal source for audio data from UPL to mobile.	**4.1.1** **OPTIONS panel** DAI Source → AUDIO GEN → PCM VALUE 1
	AUD*io*		→ When the digital instrument is selected in the Generator panel, UPL sends the signal set in the panel on the DAI . The signal 0 is output when the ananlog instrument is selected.	
	PCM1		→ The signal is the PCM value 1 specified for certain test cases, irrespective of the generator setting. The LSB is set for each second data word, all other bits are 0. Available only in operating mode "Interface DAI to MS" (CONF:DAI:INT DAIM). In operating mode "Interface DAI to SS or AUX to SS (CONF:DAI:INT DAIS|AUXS) the digital generator of UPL is always the sign al source for audio data.	

5.3 UPL used as Mobile Simulator

Command	Parameter	Basic unit	Meaning	Section
CONFigure:***DAI***:***INT****erface*	***DAIS****s*		→ UPL simulates a mobile (MS) and can be connected to a system simulator (SS). The direction of signals on the DAI is reversed as against the DAI to MS status: UPL *waits* for a reset pulse and *sends* the data clock at the beginning of data transmission. In this mode a digital link can be set up to instruments the DAI of which is permanently configured as system controller. This allows UPL to be connected for instance to the digital interface of the speech codec of a system simulator. The test control lines are switched to high impedance.	**4.1.2** **OPTIONS panel** Interface → DAI to SS → AUX to SS
	AUXS*s*		→ The second (auxiliary) audio interface is used. It has the same function as the DAI to SS interface but a different pin assignment. Pins designated "not used" in 3GPP TS 44.014 are used for this additional interface.	

Command	Parameter	Basic unit	Meaning	Section
CONFigure:***DAI***:***MOB****il*	 ***NORM*** ***SPEC****ial*		Selects the behaviour of UPL as mobile on the DAI. → UPL acts as defined by 3GPP TS 44.014 for a mobile on the DAI: UPL *waits* for a reset pulse and then *sends* the data clock at the beginning of data transmission. Sincce this mode is permanently defined, no other settings can be made under NORM. ***Note****:* *The characteristics of the NORMAL mode can also be set in the SPECIAL mode.* → If UPL is to be connected to a DAI which does not exactly correspond to 3GPP TS 44.014 specifications, the characteristics of the reset and clock signals can be modified. Available only in operating mode “Interface DAI to SS or AUX to SS" (CONF:DAI:INT DAIS|AUXS).	**4.1.2** **OPTIONS panel** Mobile → NORMAL → SPECIAL
CONFigure:***DAI***:***RES****et*	***INP****ut* ***GO***		→ UPL *waits* for the reset pulse same as in NORMAL mode. → UPL *sends* the reset pulse upon confirmation of START. Available only in setting CONF:DAI:MOB SPEC	**4.1.2** **OPTIONS panel** Reset → INPUT → START
CONFigure:***DAI***:***CLOC****k*	***INP****ut* ***INVI****nput* ***OUTP****ut* ***INVO****utput*		→ UPL *waits* for the data clock. → UPL *waits* for the *inverting* data clock: data are transferred on the falling edge and changed on the rising edge → UPL *sends* the data clock same as in NORMAL mode. → UPL *sends* the *inverting* data clock: data are transferred on the falling edge and changed on the rising edge. Available only in setting CONF:DAI:MOB SPEC	**4.1.2** **OPTIONS panel** Clock → INPUT → INPUT INV → OUTPUT → OUTPUT INV

5.4 Selecting the Generator

Command	Parameter	Basic unit	Meaning	Section
INSTrument[1][:SELect] analogous to **INST**rument[1]:**NSEL**ect	 ***A25*** ***D48*** **1** **3**		Analog and digital generator instrument → 2-channel analog generator 2 Hz to 21.75 kHz, with low-distortion generator (Option UPL-B1) 10 Hz to 110 kHz → 1-channel digital generator 2 Hz to 3.96 kHz The generator attains 99% of the Nyquist frequency. The sample frequency is 8 kHz. So the maximum generator frequency is *f_{max} = 0.5 x sample frequency x 0.99= 3.96 kHz*	**4.2.1** **GEN panel** INSTRUMENT → ANALOG → DIGITAL

5.5 Configuration of Digital Generator

Command	Parameter	Basic unit	Meaning	Section
OUTPut:***COD**ing*	 ***PCML**in* ***ALAS**imulation*		The generator signal is subjected to a coding simulation which routes the signal through a coder-decoder path. → No coding simulation, the signal is output with a 13-bit PCM linear coding. → Generator signal is A-law coded and decoded again. Available only in "INST D48"	**4.2.2** **GEN panel** Coding → PCM LINEAR → A-LAW-SIMUL
OUTPut:***SEL**ect*	 **OFF** ***CH1***		Only 1 channel is available; it can be switched on or off. → Generator switched off → Generator switched on Available only in "INST D48"	**4.2.2** **GEN panel** **Channel(s)** → OFF → 1

Command	Parameter	Basic unit	Meaning	Section
OUTPut:**SAMP**le[:FREQuency]:**MODE**	***F08***		Output clock rate. The digital generator can only be operated at a fixed clock rate of 8 kHz. The clock may be internally generated or externally applied. The *signal source* is the *same* for the digital generator and analyzer and can be selected in the OPTIONS panel by setting Interface: Setting Interface DAI to SS (CONF:DAI DAIS): internal clock generation Setting Interface DAI to MS (CONF:DAI DAIM): external clock via DAI interface	**4.2.2** **GEN panel** Sample Freq → 8 kHz
OUTPut:***AUD**iobits*	**<n>** only 13		Word width of generated audio samples in bits. The GSM DAI operates with a fix word width of 13 bits.	**4.2.2** **GEN panel** Audio Bits

5.6 Generator Functions

Command	Parameter	Basic unit	Meaning	Section
SOURce:**FUNC**tion[:SHAPe]	 ***SIN**usoid* **MULT**isine ***BURS**t* **S2P**ulse **MDIS**t **DFD** ***RAND**om* ***USER*** **POL**arity ***FSK*** ***O131***		Generator signal: → Sine → Multisine (up to 17 lines) → Sine burst → Sine^2 burst → Double sine (similar to SMPTE) → Double sine (intercarrier method) → Noise → User-defined waveform → Polarity test signal → Frequency keying → Random noise with feedback of analyzer signal; required for stability margin measurement of GSM acoustic test cases.	4.2.3 **GEN panel** FUNCTION → SINE → MULTISINE → SINE BURST → SINE^2 BURST → MOD DIST → DFD → RANDOM → ARBITRARY → POLARITY → FSK → RANDOM+ANLR

5.7 Random Noise With Feedback of Analyzer Signal

Command	Parameter	Basic unit	Meaning	Section
SOURce:**FUNC**tion[:SHAPe]	***O131***		The feedback analyzer signal whose gain is selectable is superposed on a noise signal whose frequency range can be defined. The superposed analyzer signal can either be taken from test channel1 or 2. See following command "SOUR:LOOP:CHAN OFF|CH1|CH2". This special signal can only be generated if the "Volt Range" in the GENERATOR panel is set to FIX ("SOUR:VOLT:RANG:AUTO OFF"). Otherwise, the loop signal is set to gain 0 (muting); the noise signal remains active.	**4.2.3.3** **GEN panel** FUNCTION → RANDOM+ANLR

Command	Parameter	Basic unit	Meaning	Section
SOURce:**VOLT**age[:LEVel][:AMPLitude]:**OFFS**et:**STAT**e **OFF** **ON**			DC offset allows a DC voltage to be superposed on the generator output. → Almost no DC component at output → DC component can be set by means of the following command. ***Note:*** *This setting is not possible in the analog generator if the low distortion generator is used.*	**4.2.3.3** **GEN panel** DC Offset → OFF → ON
SOURce:**VOLT**age[:LEVel][:AMPLitude]: **OFFS**et	**<n>** -5 V to 5 V -10 V to 10 V -1 Fs to 1 FS	 V V FS	Amplitude of DC component Analog instrument (OUTP:TYPE UNB) Analog instrument (OUTP:TYPE BAL) Digital instrument	**4.2.3.3** **GEN panel** DC Offset
SOURce:**RAND**om:**SPAC**ing:**MODE**	 ***USER**defined* ***ATR**ack*		Setting the frequency spacing: → Value entered is corrected to the nearest settable value (see following command). → Value of the FFT frequency spacing is transferred automatically and can be read out by means of the command CALC:TRAN:FREQ:RES? provided that FFT measurement has been selected in the analyzer.	**4.2.3.3** **GEN panel** Spacing → USER DEF → ANLR TRACK
SOURce:**RAND**om:**SPAC**ing:**FREQ**uency	**<nu>** Lower limit value: analog = 2.93 Hz digital = 0.488 Hz	Hz	Setting value for the frequency spacing for command SOUR:RAND:SPAC:MODE USER. The range of values depends on the selected generator and sampling rate. .	**4.2.3.3** **GEN panel** Spacing
SOURce:**RAND**om:**FREQ**uency:**LOW**er	**<nu>** 350 Hz to 550 Hz - 1 x Spacing	Hz	Setting the lower frequency limit for the generated noise.	**4.2.3.3** **GEN panel** Lower Freq
SOURce:**RAND**om:**FREQ**uency:**UPP**er	**<nu>** Lower Freq + 1 x Spacing to 550 Hz	Hz	Setting the upper frequency limit for the generated noise.	**4.2.3.3** **GEN panel** Upper Freq

Command	Parameter	Basic unit	Meaning	Section
SOURce:**VOLT**age:**CRES**tfactor:**MODE**	***MIN**imized* **VAL**ue		Selecting the algorithm to define the phase of each of the frequency lines and so the crest factor of the total signal. → Automatic *minimization of the crest factor* by the internal optimization of the phases. → Entering the *desired crest factor* with the following command.	**4.2.3.3** **GEN panel** Crest Fact → OPTIMIZED → VALUE
SOURce:**VOLT**age:**CRES**tfactor	**<n>** 1 to 100		The phases of the frequency lines are modified internally such that the resulting crest factor closely approximates the desired value. The accuracy of this method depends on the toatal number of lines, thus spacing and frequency range. Available only for SOUR:VOLT:CRES:MODE VAL	**4.2.3.3** **GEN panel** Crest Fact
SOURce:**VOLT**age:**TOT**al[:LEVel][:AMPLitude]	**<nu>**	V FS	Setting the peak output signal of the noise signal. The value entered here will *not affect* the loop signal. SOUR:VOLT:TOT and SOUR:VOLT:TOT:RMS are coupled via the crest factor. Therefore, if SOUR:VOLT:TOT is modified, the value of SOUR:VOLT:TOT:RMS will also change. If the crest factor is changed, SOUR:VOLT:TOT will remain unchanged. SOUR:VOLT:TOT is voltage-limited by the command SOUR:VOLT:LIM	**4.2.3.3** **GEN panel** RND PEAK
SOURce:**VOLT**age:**TOT**al:**RMS**	**<nu>**	V FS	Setting the RMS output voltage (for analog generator only). The input here will *not affect* the loop signal. SOUR:VOLT:TOT and SOUR:VOLT:TOT:RMS are coupled via the crest factor (which is constant for a particular noise signal). Therefore, if SOUR:VOLT:TOT:RMS is modified, the value of SOUR:VOLT:TOT will also change. If the crest factor is changed, SOUR:VOLT:TOT:RMS will also be changed.	**4.2.3.3** **GEN panel** RND RMS
SOURce:***LOOP**:**CHAN**nel*	**OFF** ***CH1*** ***CH2***		Specifying the analyzer channel whose signal is to be fed back to the generator and superposed onto the noise signal. → Feedback is switched off. Feedback amplification can be preset even if the loop is switched off → Channel 1 is fed back → Channel 2 is fed back	**4.2.3.3** **GEN panel** Loop Chan → OFF → 1 → 2

Command	Parameter	Basic unit	Meaning	Section
SOURce:***LOOP***:***GAIN***	**<nu>** Dimensionless factor or dB 0 to 1000		Setting the feedback gain factor ***Note:*** *As soon as "Volt Range" (on GENERATOR panel) is switched to AUTO, loop gain is automatically set to 0 by the program (muting).*	**4.2.3.3** **GEN panel** Loop Gain

5.8 Auxiliary Generator

Command	Parameter	Basic unit	Meaning	Section
SOURce**2**:**FUNC**tion	 **OFF** ***ANLG****out*		Activation of auxiliary generator → Auxiliary generator switched off; only digital audio data are generated, analog outputs are switched off → The auxiliary generator is provided at the analog outputs. Frequency and level of the analog signal can be set or swept.	**4.2.4** **GEN panel** AUX GEN → OFF → ANALOG OUT

5.9 Selecting the Analyzer

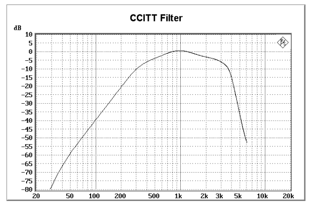
Command	Parameter	Basic unit	Meaning	Section
INSTrument**2**[:SELect] analogous to **INST**rument**2**:**NSEL**ect	***A22*** ***A110*** ***D48*** **1** **2** **4**		→ 2-channel analog analyzer DC/10 Hz to 21.9 kHz → 2-channel analog analyzer DC/20 Hz to 110 kHz → 1-channel digital analyzer DC/10 Hz to 3.656 kHz	**4.3.1** **ANLR panel** INSTRUMENT → ANLG 22 kHz → ANLG 110 kHz → DIGITAL

5.10 Configuration of Digital Analyzer

Command	Parameter	Basic unit	Meaning	Section
INPut[]:***COD****ing*	 ***PCML****in* ***ALAS****imulation*		The received signal can be subjected to a coding simulation before it is handled by the measurement routines. → No coding simulation; signal is further processed unchanged. → The signal is routed through an A-law coder-decoder path before it is measured by the analyzer.	**4.3.2** **ANLR panel** **Coding** → PCM LINEAR → A-LAW-SIMUL
INPut[]:**FILT**er[:LPASs]:**FREQ**uency	**<nu>** Query only 10 Hz	Hz	Lower frequency range limit for measurement of digital audio data.	**4.3.2** **ANLR panel** Min Freq

Command	Parameter	Basic unit	Meaning	Section
INPut[]:***SAMP**le*:**FREQ**uency:**MODE**	***F08***		Analyzer clock rate. The digital analyzer can only be operated at a fixed clock rate of 8 kHz. The clock may be internally generated or externally applied. The *signal source* is the *same* for the digital generator and analyzer and can be selected in the OPTIONS panel by setting Interface: Setting Interface DAI to SS (CONF:DAI DAIS): internal clock generation Setting Interface DAI to MS (CONF:DAI DAIM): external clock via DAI interface	**4.3.2** **ANLR panel** Sample Frq → 8 kHz
INPut[]:***AUD**iobits*	**<n>** only 13		Word width of received audio samples in bits. The GSM DAI operates with a fixed word width of 13 bits.	**4.3.2** **ANLR panel** Audio Bits

5.11 Analyzer Filters

Command	Parameter	Basic unit	Meaning	Section
SENSe[1]:**FILT**er**<i>**:***UFIL**ter1***..***UFIL**ter9*[:STATe]	**<i>***) = 1 to 3 **ON|OFF**		A HPASs, LPASs, BPASs, BSTOp, NOTCh, TERZ, OCTav or FILE filter can be assigned to each of the 9 user filters (UFILter) whose parameters are freely selectable (see SENSe:FILTer<n>:HPASs|:LPASs ... and following commands). When the filter is switched ON the previously active filter is switched OFF.	**4.4** **FILTER panel** Filter
SENSe[1]:**FILT**er**<i>**:**CCIT**t[:STATe]	**<i>***) = 1 to 3 **ON|OFF**		Filter: CCITT Standards: CCITT 0.41 IEEE Rec. 743-84 CISPR 6-76 CCITT Rec. P.53 	**4.4** **FILTER panel** Filter → CCITT

5.12 Fast Switch-Off of Outputs

Command	Parameter	Basic unit	Meaning	Section
OUTPut	**OFF** **ON**		→ The digital output is set to OFF(0 FS). Analog outputs are terminated (impedance is retained); output level = 0 V. → Switched-off lines can be reactivated.	**4.5** Key OUTPUT OFF

5.13 MACRO Operating

Command	Parameter	Basic unit	Meaning	Section
SYSTem:***PROG****ram*:***EXEC****ute*	**'filename'**		This command allows to load and start any BASIC program with the name <filename> (preferred file extension: *.BAS). After completion of the program, a 1 → 0 transition is generated in the "RUN" bit (#14) of the operation register. The controller receives this information by SRQ or serial poll and can fetch the measurement data. The data exchange between the external control program and the BASIC program can be performed via the result display, the result buffer or the block data input/output by means of the following command "SYST:PROG **<n>**{,**<n>**}". Starting a BASIC macro with this command is only possible via IEC/IEEE-bus or RS232 remote control. A program of the Universal Sequence Controller UPL-B10 **cannot** start a BASIC macro! **For detailed program example see Appendix C.**	**2.16** **OPTIONS panel** Exec Macro <filename>
SYSTem:***PROG****ram*[:DATA]	**<n>**{,**<n>**}		From the BASIC macro, up to 1024 floating-point values can be transferred to the external control program. The BASIC macro enters the values into the block buffer which are then read out by the external control program.	**No manual control**
SYSTem:***PROG****ram*:**POIN**ts?	**<n>** 0 to 1024 Query only		Number of available block data values that were entered into the block buffer by the BASIC macro.	**No manual control**

5.14 Commands for Data Output

Command	Parameter	Basic unit	Remark	Section
SENSe[1]:**DATA**[1|**2**]?	**<nu>**	Depen .on FUNC	Returns the measured value of the first analyzer for functions RMS, RMSS, PEAK, QPE, DC, THD, THDN, MDIST, DFD and WAF. DATA1 selects input channel 1 DATA2 selects input channel 2.	**3.15.8** **Result display**
SENSe**2**:**DATA**[1|**2**]?	**<nu>**	V/FS	Returns the measured value of the second analyzer (peak voltmeter). DATA1 selects input channel 1 DATA2 selects input channel 2.	**3.15.8** **Result display**
SENSe**3**:**DATA**[1|**2**]?	**<nu>**	Hz	Returns the measured value of the third analyzer (frequency counter). DATA1 selects input channel 1 DATA2 selects input channel 2.	**3.15.8** **Result display**
SENSe**4**:**DATA**[1]?	**<nu>**	DEG	Returns the measured value of the phase and group-delay measurement.	**3.15.8** **Result display**

Universal Sequence Controller (UPL-B10), IEC/IEEE-bus or RS232 remote control also allow write access to the result buffers.
This is of particular importance for the BASIC-macro mode:

- Measured values calculated by a BASIC macro can be indicated to the user in the usual result windows.
  Any floating-point parameters and measured values can be exchanged between the BASIC macro and the controller via the result buffer.

5.15 Calling test cases

With the UPL, setting and measurement sequences can be written as BASIC programs or recorded using the built-in program generator (see 3.15.3 Command Logging-Converting B10 into IEC/IEEE-Bus Commands). Option UPL-B10 (Universal Sequence Controller) is required. The generated BASIC programs can be stored (preferred file extension: .BAS) and called and used in various ways (see Macro-Operation).

The following example illustrates how a BASIC macro is called by means of an IEC/IEEE-bus control program in the programming language C and the IEC/IEEE-bus driver GPIB.COM from National Instruments:

IEC/IEEE-bus program in the controller calling test case TC301.BAS in the UPL:

```
/***************************************************************************
* Simple program example in C for calling a test case in the UPL using the
* GPIB driver of NATIONAL INSTRUMENTS.
****************************************************************************/
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include <conio.h>
#include "C:\NI-GPIB\C\DECL.H"

#define UPL_IEC_ADR 20

/***************************************************************************
* Error output
****************************************************************************/
void report_error(int fd, char *errmsg)
{
 fprintf(stderr, "Error %d: %s\n", iberr, errmsg);
 if (fd != -1)
  {
   printf("Switch off remote control\n");
   ibonl(fd,0);
  }
getch();
exit(1); // Program abort
}

/***************************************************************************
* Output of IEC/IEEE-bus command
****************************************************************************/
void befout (int upl, char *befstr)
{
 ibwrt(upl, befstr,(long)strlen(befstr));
 if (ibsta & ERR)
  report_error (upl, "No command output to UPL!\n");
}

/***************************************************************************
* Read-in of query response
****************************************************************************/
void queryin (int upl, char *reading)
{
 ibrd(upl, reading, 100L);
  if (ibsta & ERR)
   report_error (upl, "No data received from UPL!\n");
  reading[ibcnt-1] = '\0'; // Overwrite line feed with string terminator
```

```
}

/**************************************************************************
* Main program
**************************************************************************/
int main()
{
 int upl;
 char iec_bef [80];
 long count = 0;
 char stb;
 char testcase[80];
 char response [80];

 DevClear(0, UPL_IEC_ADR); // 0 = Board number

 if ((upl = ibdev(0, UPL_IEC_ADR, 0, T10s, 1, 0)) < 0)
  report_error (upl, "UPL cannot be initialized!\n");

 ibconfig (0,IbcAUTOPOLL, 1); // Enabling polling

 befout (upl,"*ESE 0;*SRE 0");        // Disables ESR and SRQ
 befout (upl,"STAT:OPER:ENAB 16384"); // Enables bit 14 (RUN BASIC macro) in
                                      // the OPERation register
 befout (upl,"STAT:OPER:NTR 16384");  // 1->0-0 transition on bit 14
 befout (upl,"STAT:OPER:PTR 0");      // 0->1-1 transition of bit 14


// Before starting the BASIC macro, the OPER bit in the STB
// must be cleared. This is done by reading out the EVENt bit of the
// OPERation register.

 befout (upl,"STAT:OPER:EVEN?");
 queryin (upl,response); // Readout without processing

 strcpy (testcase,"C:\\GSM\\TC301.BAS"); // Name of test case
 sprintf (iec_bef,"SYST:PROG:EXEC ’%s’",testcase);
 befout (upl,iec_bef);        // Starts test case
 stb = 0;                     // Wait until test case is completed.
 while ((stb & 0x80) == 0)    // Performs a serial poll
  {                           // until bit 7 (OPER) of the STB is 1.
   ibrsp (upl,&stb);
   if ((count++ % 100) == 0) // Progress indication as long as ...
    printf ("_");             // ... the OPER bit in the STB is waited for */
  }
 printf ("\n>>>>>>>>>> Testcase %s of test case executed! <<<<<<<<<",testcase);

 ibonl(upl, 0); // Switches off remote control

 return (0);
}
```

6 Default Setup of UPL Model 16

The extensions of UPL16 compared to the basic model of UPL06 are described in the following. Unchanged settings are described in the operating manual of UPL.

The default setup of UPL is effected with the following settings:

Manual FILE-Panel:	IEC/IEE bus:
LOAD INSTRUMENT STATE Mode DEF SETUP	*RST

6.1 Default Setup of Generators

INSTRUMENT ANALOG

The following is valid for setting GENERATOR → ANALOG (default setting):

```
. Channel(s) 2 = 1
. Output     UNBAL
. Volt Range AUTO
. Max Volt   12.000 V
. Ref Freq   1000.0 Hz
. Ref Volt   1.0000 V
```

The following is valid for setting GENERATOR → DIGITAL:

. Coding	PCM LINEAR	Further selection point: A-LAW SIMUL
. Channel(s)	1	OFF or 1 can be selected
. Sample Frq	8 kHz	Fixed at 8 kHz
. Audio Bits	13	Fixed at 13 bit
. Max Volt	1.0000 FS	
. Ref Freq	1000.0 Hz	
. Ref Volt	1.0000 FS	

```
. AUX GEN     OFF
```

The following is valid for setting AUX GEN → ANALOG OUT:

```
. Channel(s)  2 = 1
. Output      UNBAL
. SWEEP CTRL  OFF
. Anlg Freq   1000.0 Hz
. Anlg Ampl   0.1000 V
```

The auxiliary generator (AUX GEN) has its own sweep system which is configured in the same way as the sweep system of the functional generator. A 2-dimensional sweep (Z-axis), ie frequency and level at the same time, is not implemented..

Generator function RANDOM+ANLR

FUNCTION	RANDOM+ANLR	
· DC Offset	OFF	ON: 0.0000 FS or· 0.0000 V
. Spacing	USER DEF	Further selection point: ANLR TRACK
	10.000 Hz	Not for spacing ANLR TRACK
. Lower Freq	350.00 Hz	
. Upper Freq	550.00 Hz	
. Crest Fact	OPTIMIZED	Further selection point: VALUE:
	1.0000	For Crest Fact VALUE:
. RND PEAK	1.0000 V	1.0000 FS in digital generator
. RND RMS	0.3869 V	**0.4657** FS in digital generator
. Loop Chan	1	
. Loop Gain	0.0000 *	

6.2 Default Setup of Analyzers

INSTRUMENT — ANLG 22kHz

The following is valid for setting ANALYSATOR ANLG 22 kHz and ANLG 110 kHz:

```
. Min Freq    10 Hz          20 Hz for ANLG 110kHz
. Ref Imped   600.00 Ω
. Channel(s)  1
. Ch1 Coupl   AC
. Ch1 Input   BAL
. Ch1 Imped   200 kΩ
. Ch1 Common  FLOAT          with channel 2 selected, the same settings apply
. Ch1 Range   AUTO
```

The following is valid for setting ANALYSATOR DIGITAL:

```
. Coding      PCM LINEAR     Further selection point: A-LAW SIMUL
. Min Freq    10 Hz
. Sample Frq  8 kHz          Fixed at 8 kHz
. Audio Bits  13             Fixed at 13 bit
```

6.3 Default Setup of Option Panel

```
. Remote via  IEC BUS        If remote option (UPL-B4) is installed
. UPL IECadr  20             Not for Remote via COM2
. Exec Macro  BASIC FILE
. Beeper      ON

PARAM.LINK  ———————————
. Param Link  CHOICE...      Function tracking Gen → Anl is selected

DIGITAL I/O CONFIG ——
. Interface   DAI to MS      Further selection points: DAI to SS | AUX to SS
. Reset       START
. Test Ctrl   NORMAL MODE    Further selection points:SPEECH DEC|SPEECH ENC|ACOUST DEVS
. DAI Source  AUDIO GEN      Further selection point: PCM VALUE 1
```

The following is valid for setting Interface DAI to SS or AUX to SS:

```
. Mobile      NORMAL         Further selection point: SPECIAL
```

The following is valid for setting Mobile SPECIAL:

```
. Reset       START          Further selection point: INPUT
. Clock       INPUT          Further selection points: INPUT INV | OUTPUT | OUTPUT INV.
  Clock       INPUT          Further selection points: INPUT INV | OUTPUT | OUTPUT INV
```

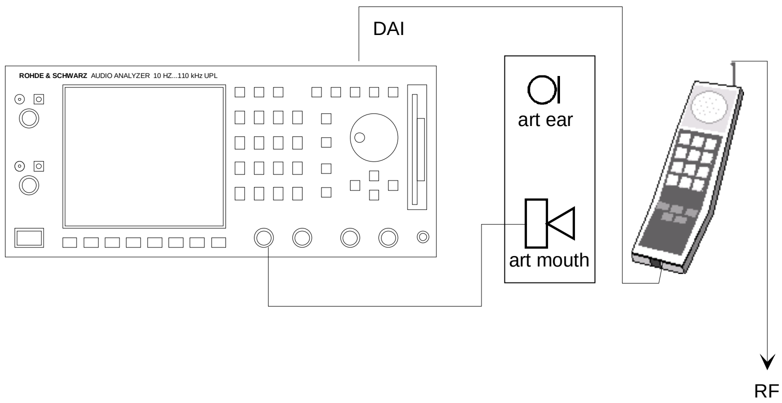
Annex A: Acoustic Testcases for GSM Mobiles

30.1 Sending sensitivity/frequency response

30.1.1 Definition and applicability

The sending sensitivity frequency response is, as a function of the input test tone frequency, the ratio expressed in dB between the output level, represented by the PCM bit stream at the Digital Audio Interface (DAI) and the input sound pressure in the artificial mouth required to obtain this.

Block diagram



30.1.4 Method of test

30.1.4.1 Initial conditions

a) The handset is mounted in the LRGP (see annex A of ITU-T Recommendation P.76). The earpiece is sealed to the knife-edge of the artificial ear.

b) A pure tone with a sound pressure of -4,7 dBPa (in accordance with ITU-T Recommendation P.64) is applied at the mouth reference point (MRP) as described in ITU-T Recommendation P.64 using an artificial mouth conforming to ITU-T Recommendation P.51.

c) A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, is connected to the Digital Audio Interface (DAI). The DAI is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.1.4.2 Procedure

The SS measures the output level represented by the PCM bit stream at the DAI (pin 23) at one-twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 000 Hz inclusive.

30.1.5 Test requirement

The sending sensitivity/frequency response (from MRP to the DAI) shall be within a mask given in table 30.1. The mask can be drawn with straight lines between the breaking points in the table on a logarithmic (frequency) vs linear (dB sensitivity) scale.
All sensitivity levels are dB on an arbitrary scale.

Table 30.1

Frequency (Hz)	**Upper Limit (dB)**	**Lower Limit (dB)**
100	-12	
200	0	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	0	

The used frequencies in Hz are stored in the file GSM.SPF, sub-multiples of 8 kHz are avoided according to Note 2 in chapter 30 of 3GPP TS 51.010-1:

101	212	450	950	2002
106	224	475	1002	2120
112	236	502	1060	2240
118	251	530	1120	2360
126	265	560	1180	2500
132	280	600	1250	2650
140	300	630	1320	2800
150	315	670	1400	3000
161	335	710	1500	3150
170	355	750	1602	3350
180	375	802	1700	3550
190	402	850	1800	3750
201	425	900	1900	3950

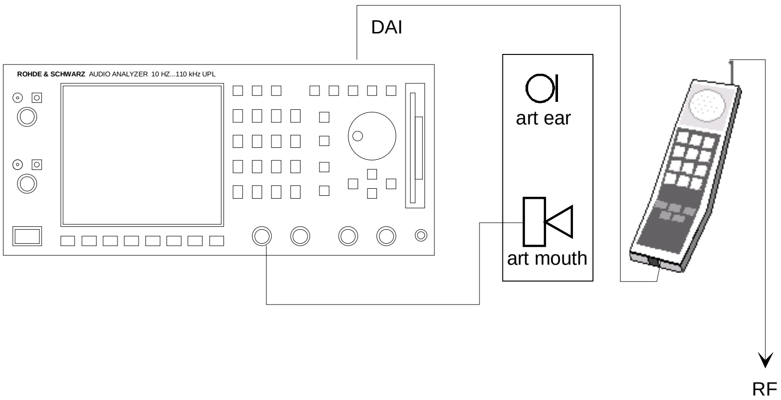
The frequency response is measured and the limit files 301.LUP and 301.LLW are invoked. Those files contain the limit values for sending frequency response according to Table 30.1 of 3GPP TS 51.010-1. The maximum deviation in upper and lower direction to the initial limit values are calculated and the measured curve is then shifted by the mean value of max upper and max lower deviation. After that a new limit calculation is done. If the shifted frequency response is within the limit lines PASS is displayed, when it is outside FAIL is displayed.

30.2 Sending loudness rating

30.2.1 Definition and applicability

The Sending Loudness Rating (SLR) is a means of expressing the sending frequency response based on objective single tone measurements in a way which relates to how a speech signal would be perceived by a listener.

Block diagram



30.2.4 Method of test

30.2.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

30.2.4.2 Procedure

a) The sending sensitivity is measured at each of the 14 frequencies given in table 2 of ITU-T P.79, bands 4 to 17.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR is calculated according to ITU-T Recommendation P.79 formula 4.19 b of ITU-T P.79, over bands 4 to 17, using the sending weighting factors from ITU-T Recommendation P.79 table 2, adjusted according to table 3 of ITU-T Recommendation P.79.

Frequencies used are stored in File P79FREQ in Hz, used values No.4 to 17:

201	1002
251	1250
315	1602
402	2002
502	2500
630	3150
802	3950

The weighting factors according to table 2 in ITU-T P79 are stored in file P79WS and are read out by the program.

The measured values are converted to levels in dBV according to following calculation:

The digital level is measured via the DAI in dBFS

0 dBFS corresponds to 3.14 dBm0 ´=` 1.11193 V ´=` 0.9216 dBV this leads to the correction formula:

Level in dBV = measured value in dBFS + 0.9216 dB

The sensitivity has to be expressed in dBV/Pa. The level is measured at an acoustic sound pressure level of -4.7 dBPa so the sensitivity in dBV/Pa equals :

Sensitivity in dBV/Pa = measured value in dBFS + 0.9216 dB + 4.7 dB

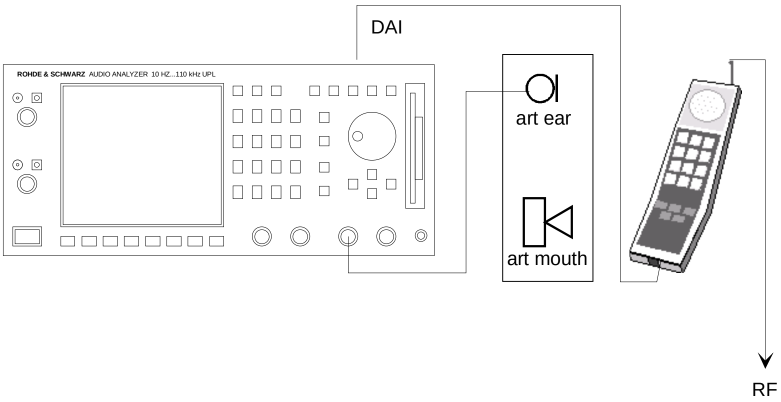
This sensitivity is measured for each frequency, the sending loudness rating is then calculated according to formula 4.19 b of ITU-T P79 and corrected by -0.3 dB according to table 3 of ITU-T P79

30.3 Receiving sensitivity/frequency response

30.3.1 Definition and applicability

The receiving sensitivity frequency response is, as a function of the input test tone frequency, the ratio expressed in dB between the output sound pressure in the artificial ear and the input level, represented by the PCM bit stream at the Digital Audio Interface (DAI), required to obtain this.

Block diagram



30.3.4 Method of test

30.3.4.1 Initial conditions

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

b) A digital signal generator is connected at the digital interface delivering a signal equivalent to a pure tone level of -16 dBm0, see ITU-T Recommendation P.64.

c) The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.3.4.2 Procedure

Measurements are made at one twelfth-octave intervals as given in the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. At each frequency, the sound pressure in the artificial ear is measured by connecting a suitable measuring set to the artificial ear.

30.3.5 Test requirement

The receiving sensitivity/frequency response (from the DAI to the ERP) shall be within the mask given by table 30.2. The mask can be drawn with straight lines between the breaking points in the following table on a logarithmic (frequency) vs linear (dB sensitivity) scale.
All sensitivity levels are dB on an arbitrary scale.

Table 30.2

Frequency (Hz)	Upper Limit (dB)	Lower Limit (dB)
100	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	2	

NOTE: * The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) vs linear (dB) scale.

The used frequencies in Hz are stored in the file GSM.SPF, sub-multiples of 8 kHz are avoided according to Note 2 in chapter 30 of 3GPP TS 51.010-1:

101	212	450	950	2002
106	224	475	1002	2120
112	236	502	1060	2240
118	251	530	1120	2360
126	265	560	1180	2500
132	280	600	1250	2650
140	300	630	1320	2800
150	315	670	1400	3000
161	335	710	1500	3150
170	355	750	1602	3350
180	375	802	1700	3550
190	402	850	1800	3750
201	425	900	1900	3950

Using artificial ear ITU-T Recommendation P.57 Type 1:

The frequency response is measured and the limit files 303.LUP and 303.LLW are invoked. Those files contain the limit values for receiving frequency response according to Table 30.2 of 3GPP TS 51.010-1. The maximum deviation in upper and lower direction to the initial limit values are calculated and the measured curve is then shifted by the mean value of max upper and max lower deviation. After that a new limit calculation is done. If the shifted frequency response is within the limit lines PASS is displayed, when it is outside FAIL is displayed.

Using artificial ear ITU-T Recommendation P.57 Type 3.2:

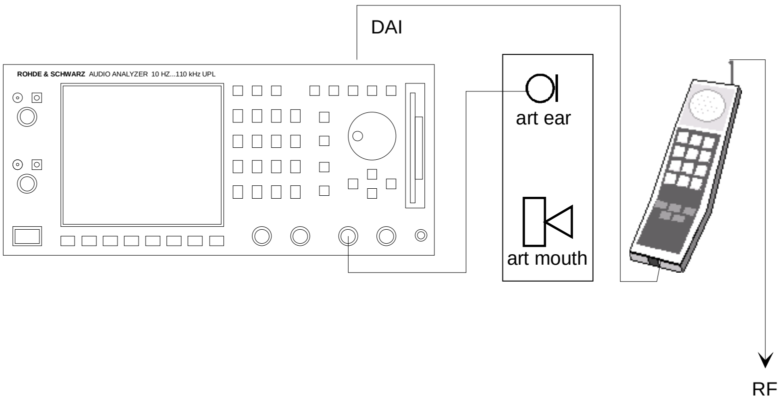
The frequency response is measured and each measurement point is corrected by the individual calibration value of the artificial ear at the specific frequency using calibration file EAR_32L.CAL. After correction the limit files 303.LUP and 303.LLW are invoked. Those files contain the limit values for receiving frequency response according to Table 30.2 of 3GPP TS 51.010-1. The maximum deviation in upper and lower direction to the initial limit values are calculated and the measured curve is then shifted by the mean value of max upper and max lower deviation. After that a new limit calculation is done. If the shifted frequency response is within the limit lines PASS is displayed, when it is outside FAIL is displayed.

30.4 Receiving loudness rating

30.4.1 Definition and applicability

The Receiving Loudness Rating (RLR) is a means of expressing the receiving frequency response based on objective single tone measurements in a way which relates to how a speech signal would be perceived by a listener.

Block diagram



30.4.4 Method of test

30.4.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

30.4.4.2 Procedure

a) The receiving sensitivity is measured at each of the 14 frequencies listed in table 2 of ITU-T Recommendation P.79, bands 4 to 17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR is calculated according to ITU-T Recommendation P.79 formula 4.19 c, over bands 4 to 17, using the receiving weighting factors from table 2 of ITU-T Recommendation P.79, adjusted according to table 3 of ITU-T Recommendation P.79.

c) The artificial ear sensitivity must be corrected according to the real ear correction of table 4 of ITU-T Recommendation P.79.

NOTE:
The values of real ear correction in ITU-T Recommendation P.79 table 4 were derived for one type of handset conforming to the shape defined in ITU-T Recommendation P.35. These values are used in this ETS because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change this ETS to use the values appropriate to each handset.

Frequencies used are stored in File P79FREQ in Hz, used values No.4 to 17:

201	1002
251	1250
315	1602
402	2002
502	2500
630	3150
802	3950

The weighting factors according to table 2 in ITU-T P79 are stored in file P79WR and are read out by the program.

The leakage correction factors according to table 4 in ITU-T P79 are stored in file P79LE and are read out by the program.

Calculation of RLR:

The digital level is generated via the DAI in dBFS

0 dBFS corresponds to 3.14 dBm0 ´=` 1.11193 V ´=` 0.9216 dBV this leads to the correction formula:

0 dBV corresponds to 0 dBFS - 0.9216 dB

Source level in dBFS = value in dBm0 -3.14 dB -16 dBm0 ´=` -19.14 dBFS

The analyzer measures the sound pressure level in the ear in dBPa.

The sensitivity has to be expressed in dBPa/V. The generator level is set to -19.14 dBFS so the sensitivity in dBPa/V equals :

Sensitivity in dBPa/V = measured value in dBPa + 19.14 dB - 0.9216 dB

This sensitivity is measured for each frequency, the receiving loudness rating is then calculated according to formula 4.19 c of ITU-T P79 and corrected by -0.3 dB according to table 3 of ITU-T P79

Depending on the volume setting the RLR has to be within 2 ± 3 dB at nominal setting of volume control or ≥ -13 dB at maximum volume setting.

Using artificial ear ITU-T Recommendation P.57 Type 3.2:

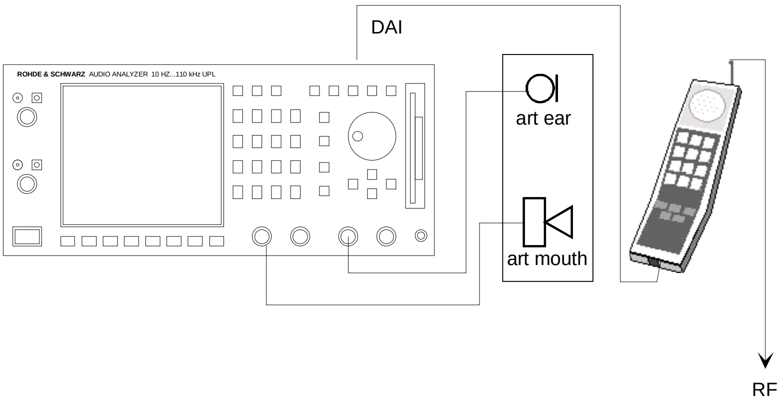
Each measured value is corrected by the individual correction data of the artificial ear using calibration file EAR_32L.CAL and no leakage correction is made so values of table 4 in ITU-T P79 are not used.

30.5.1 Side Tone Masking Rating (STMR)

30.5.1.1 Definition and applicability

The sidetone loudness ratings are a means of expressing the path loss from the artificial mouth to the artificial ear based on objective single tone measurements in a way that relates to how a speaker will perceive his own voice when speaking (talker sidetone, expressed by the sidetone masking rating - STMR), or how a listener will perceive the background noise picked up by the microphone (listener sidetone rating - LSTR).

Block diagram



30.5.1.4 Method of test

30.5.1.4.1 Initial conditions

a) The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A".

b) The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51.

30.5.1.4.2 Procedure

a) The SS sends a PCM bit stream coded with the value No 1 over the DAI (pin 25). Or alternatively the activation of the A/D and D/A converters is performed via a call setup, in which case the DAI connection between the MS and SS, and the PCM bit stream are optional.
NOTE: The idle channel noise in the receiving direction is the acoustic sound pressure in the artificial ear when the digital input signal at the DAI is the PCM coded value No. 1.

b) The SS applies a pure tone with a sound pressure of -4,7 dBPa at the mouth reference point as described in ITU-T P.64 using an artificial mouth conforming to ITU-T P 51.

c) For each frequency given in table 2 of ITU-T P.79, bands 4 to 17, the sound pressure in the artificial ear is measured.

d) The sidetone path loss (LmeST) is expressed in dB and the STMR (in dB) is calculated from the formula 8.4 of ITU-T Recommendation P.79, using the weighting factors of column (3) in table 6 of ITU-T Recommendation P.79 (unsealed), and values of LE in accordance with table 4 of ITU-T Recommendation P.79.

Frequencies used are stored in File P79FREQ in Hz, used values No.4 to 17:

201	1002
251	1250
315	1602
402	2002
502	2500
630	3150
802	3950

The weighting factors according to table 6 column 3 in ITU-T P79 are stored in file P79WML and are read out by the program.

The leakage correction factors according to table 4 in ITU-T P79 are stored in file P79LE and are read out by the program.

Calculation of STMR:

The analyzer measures the sound pressure level in the ear in dBPa.

The sidetone path loss LmeST has to be expressed in dB. The level is measured at an acoustic sound pressure level of -4.7 dBPa so the sidetone path loss equals :

LmeST = -1 x (measured value in dBPa + 4.7 dB)

This sidetone path loss is measured for each frequency, STMR is then calculated according to formula 8.4 of ITU-T P79.

STMR has to be within 13 $\pm$ 5 dB at nominal setting of volume control.

Using artificial ear ITU-T Recommendation P.57 Type 3.2:

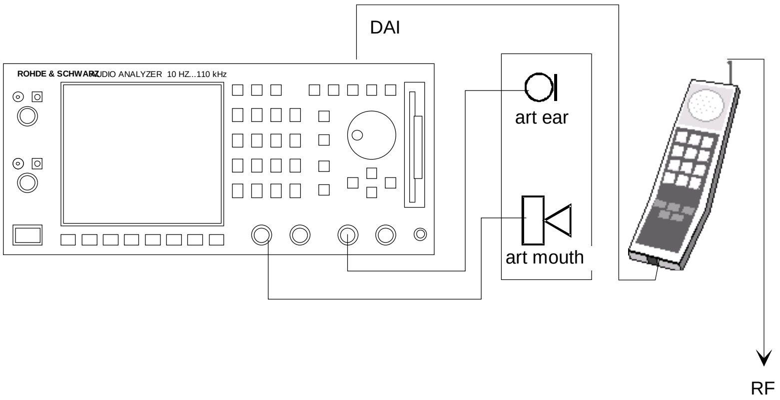
Each measured value is corrected by the individual correction data of the artificial ear using calibration file EAR_32L.CAL and no leakage correction is made so values of table 4 in ITU-T P79 are not used.

30.5.2 Listener Side Tone Rating (LSTR)

30.5.2.1 Definition and applicability

The Listener Sidetone Rating (LSTR) is considered a major parameter affecting the user perception of the system. The requirements and this test is applicable to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.5.2.4 Method of test

30.5.2.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A". The SS sends a PCM bit stream coded with the value No. 1 over the DAI (pin 25) to the MS.

30.5.2.4.2 Procedure

a) The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

b) A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Pink noise" as described in ITU-T recommendation P.64 annex B to within +/-1 dB and the level shall be adjusted to 70 dBA (-24 dBPa(A)). The tolerance on this level is +/-1 dB.

c) The artificial mouth and ear are placed in the correct position relative to MRP, the handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

d) Measurements are made in one-third octave bands for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

e) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the ITU-T Recommendation P.79 formula 8-4, using the weighting factors in column (3) in table 6 of the Recommendation, and the values of LE; in accordance with table 4 of the Recommendation.

The weighting factors according to table 6 column 3 in ITU-T P79 are stored in file P79WML and are read out by the program.

The leakage correction factors according to table 4 in ITU-T P79 are stored in file P79LE and are read out by the program.

Calculation of LSTR:

The analyzer measures the sound pressure level in the ear in dBPa in the third octave bands 4 to 17.

The sound field is adjusted to an total A-weighted sound pressure level of 70 dBA. Due to pink noise the level in each 1/3-octave band must be equal. In order to calculate the listener side tone path loss in each 1/3-octave band the level of each single 1/3-octave band has to be calculated.

Using weighting factors K1 to K14 for the A-curve weighting in the 1/3-octave bands and the Level L of a single 1/3-octave band the total sound pressure level is:

$Ltot(A) = L \times SQR(K1^2 + K2^2 + ... K14^2)$

or the level of a single 1/3-octave band is:

$L = Ltot(A) / SQR(K1^2 + K2^2 + ... K14^2)$

The correction factor $SQR(K1^2 + K2^2 + ... K14^2)$ can be calculated out of the nominal values of correction values for curve A according to publication IEC 179 at the mean frequency of each 1/3-octave band, this leads to the corection:

L = Ltot(A) / 3.373 or

L = Ltot(A) - 10.56 dB

The listener sidetone path loss LmeST has to be expressed in dB. The total A-weighted sound pressure level of the noise field is adjusted to 70 dB which corresponds to -24 dBPa, the level of each 1/3-octave band is therefore:

L = -24 - 10.56 dBPa = -34.56 dBPa

so the listener sidetone path loss equals :

LmeST = -1 x (measured value in dBPa + 34.56 dB)

This listener sidetone path loss is calculated for third octave bands 4 to 17 (centered at 200 Hz to 4000 Hz), LSTR is then calculated according to formula 8-4 of ITU-T P79.

LSTR shall not be less than 15 dB.

Using artificial ear ITU-T Recommendation P.57 Type 3.2:

Each measured value is corrected by the individual correction data of the artificial ear using calibration file EAR_32L.CAL and no leakage correction is made so values of table 4 in ITU-T P79 are not used.

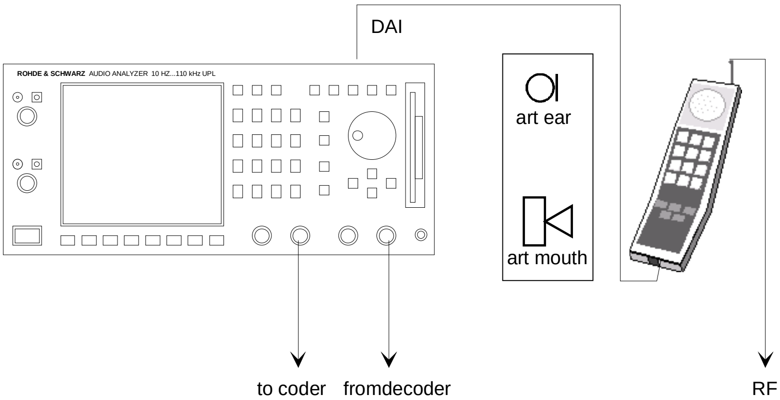
30.6.1 Echo Loss (EL)

This testcase 30.6.1 is valid up to Rel-99. From Rel-4 it is replaced by 30.17.1 (according to 3GPP TS 26.132) which is available with Upgrade UPL-U81.

30.6.1.1 Definition and applicability

The echo loss is the path loss from the input of the reference speech encoder of the SS to the output of the reference speech decoder of the SS. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.6.1.4 Method of test

30.6.1.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Normal operation". The SS sets up a speech call according to the generic call set up procedure. The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51.
Where a user controlled volume control is provided it is set to maximum.

30.6.1.4.2 Procedure

An implementation of the ITU-T P.50 artificial speech is connected to the analogue or digital input of the reference speech encoder of the SS. This implementation is either a real time algorithm producing the artificial speech or a pre-recorded tape of artificial speech. Both "male" and "female" artificial speech is required.

A ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the input signal is measured. The echo loss signal is not measured at this stage as the first ten second segment is used to allow any acoustic echo cancellation devices within the MS to adapt to the echo path.
Immediately after a second ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the echo signal is measured at the digital output of the reference speech decoder of the SS.
The difference between the third octave input power and the third octave output power is entered into the ITU-T G.122 TCL algorithm and the acoustic echo loss calculated.

The test is repeated with the "female" artificial speech and the results of both "male" and "female" averaged to give the final result.

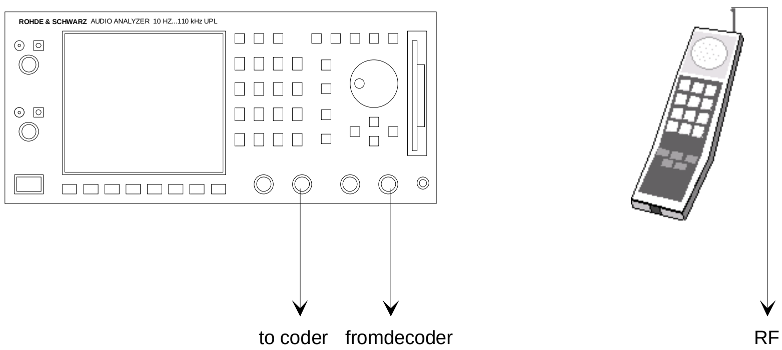
Prior to measurement of Echo Loss the speech encoder / decoder path has to be calibrated. The calibrating procedure uses the same artificial voices but the test mobile is set to the mode “echo back”. In this mode the digital audio signal fed to the mobile via the RF is sent back thus simulating an echo loop with 0 dB gain. The peak output voltage of the artificial voice generator is set to equal to full scale of the speech encoder input. Thus under this loop condition the actual gain of the encoder / decoder loop is measured and stored as reference for the echo loss measurement. This gain value is stored on file and will be used for all subsequent measurements. Normally this calibration has only to be repeated if there were any changes in hardware or used instruments, the type of test mobile should not influence this calibration.

30.6.2 Stability margin

30.6.2.1 Definition and applicability

The receive-transmit stability margin is a measure of the gain that would have to be inserted between the go and return paths of the reference speech coder in the SS for oscillation to occur.
The requirements and this test apply to all types of GSM 900 and DCS 1 800 MS supporting speech.

Block diagram



30.6.2.4 Method of test

30.6.2.4.1 Initial conditions

For handset operation the handset is placed on a hard plane surface with the transducers facing the surface. For handsfree operation the test setup is shown in ITU-T P.34 (Fig 3/ITU-T P.34), but omitting the test table.
Where a user controlled volume control is provided it is set to maximum.

30.6.2.4.2 Procedure

a) A gain equivalent to the minimum stability margin is inserted in the loop between the go and return paths of the reference speech coder in the SS and any acoustic echo control is enabled.

b) A test signal according to ITU-T O.131 is injected into the loop at the analogue or digital input of the reference speech codec of the SS and the stability is measured. The test signal has a level of -10 dBm0 and a duration of 1 s.

No audible oscillation shall be detected

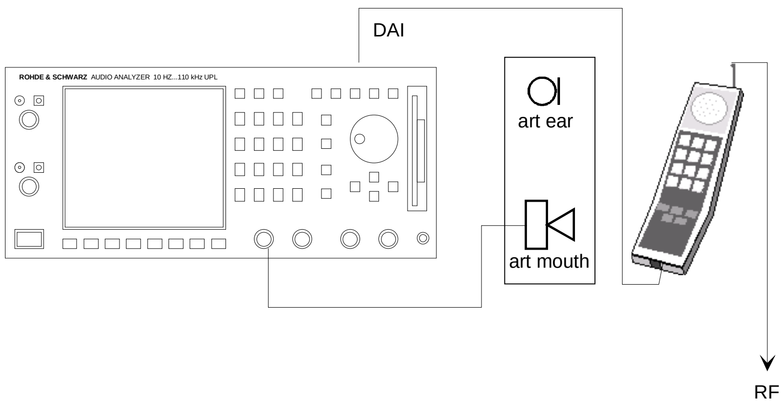
30.7 Distortion

30.7.1 Sending

30.7.1.1 Definition and applicability

The transmit signal to total distortion ratio is a measure of the linearity of the transmitter equipment (excluding the speech codec). The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.7.1.4 Method of test

30.7.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51. The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.7.1.4.2 Procedure

a) A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz is applied to the MRP. The level of this signal is adjusted until the PCM bitstream at the DAI output (pin 23) corresponds to -10 dBm0. The level of the signal at the MRP is then the acoustic reference level (ARL).

b) The test signal is applied at the following levels: -35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to the ARL.

c) The ratio of signal to total distortion power is measured at the DAI with the psophometric noise weighting (see ITU-T G.714 and O.132) at each signal level.

NOTE: *The measurement is not to be carried out at sound pressures exceeding +10 dBPa.*

Used frequency: 1015 Hz

The digital level is measured via the DAI in dBFS

0 dBFS corresponds to 3.14 dBm0

Level in dBm0 = measured value in dBFS + 3.14 dB

The sound pressure level at the MRP is adjusted until the measured value equals - 10 dBm0 which corresponds to - 13.14 dBFS at the digital input

A sweep from sound pressurel level - 35 dB up to + 10 dB relative to the acoustic reference level determined before is started with measurement function SINAD with weighting filter according to ITU-T O.132 switched on

The limit line according to table 30.3 of 3GPP TS 51.010-1 is loaded via the program and limit check is switched on

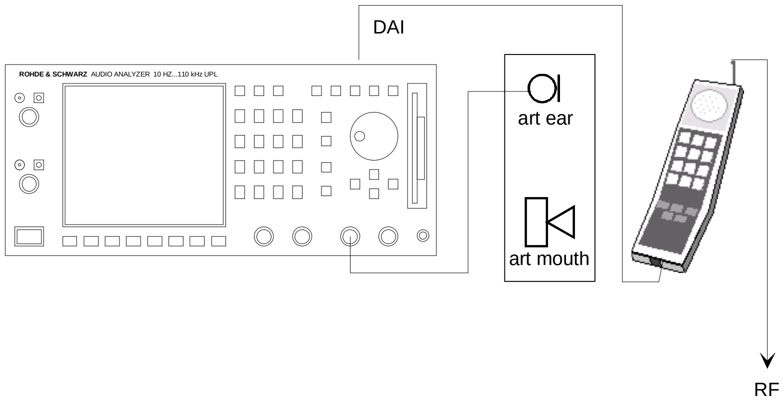
PASS is displayed if the measured curve is above the limit line otherwise FAIL is displayed

30.7.2 Receiving

30.7.2.1 Definition and applicability

The receive signal to total distortion ratio is a measure of the linearity in the receive equipment (excluding the speech decoder). The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.7.2.4 Method of test

30.7.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51. The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.7.2.4.2 Procedure

a) The SS sends, via the DAI (Pin 25), a PCM bit stream simulating a sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz corresponding to ITU-T O.132 at the following levels: -45, -40, -35, -30, -25, -20, -15, -10, -5, 0 dBm0.

b) The ratio of signal to total distortion power is measured with the psophometric noise weighting in the artificial ear (see ITU-T G.714 and O.132) at each signal level.

c) The measurement is only carried out at sound pressures between -50 dBPa and +10 dBPa.

Used frequency: 1015 Hz

The digital generator signal on the DAI is set in dBr

1 FS corresponds to 3.14 dBm0

0 dBm0 corresponds to 0.696627 FS, this is the reference level set in the generator

The acoustic sound pressure level is measured in the artificial ear.

A sweep from - 45 dB up to + 0 dB relative to the reference of 0 dBm0 is started with measurement function SINAD with weighting filter according to ITU-T O.132 switched on

The limit line according to table 30.4 of 3GPP TS 51.010-1 is loaded via the program and limit check is switched on

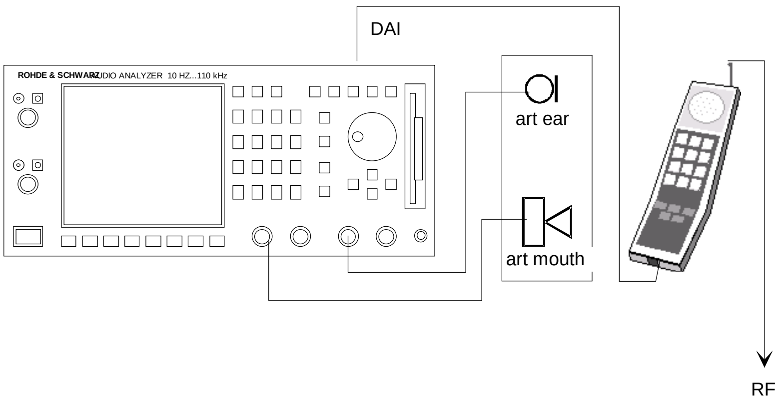
PASS is displayed if the measured curve is above the limit line otherwise FAIL is displayed

30.8 Sidetone distortion

30.8.1 Definition and applicability

The sidetone distortion expresses the linearity of the sidetone path in the handset. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.8.4 Method of test

30.8.4.1 Initial conditions

The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A". The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51.

30.8.4.2 Procedure

a) The SS sends the PCM bit stream coded with the value No 1 over the DAI (pin 25) to the MS.

b) An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range 315 Hz to 1000 Hz is connected to the artificial ear.

c) A pure-tone signal of -4,7 dBPa is applied at the mouth reference point at frequencies of 315 Hz, 500 Hz, and 1 000 Hz. For each frequency the third harmonic distortion is measured in the artificial ear.

30.8.5 Test requirement

The third harmonic distortion generated shall not be greater than 10 %.

Procedure:

For each frequency the third harmonic distortion of the side tone signal is measured and compared to the limit of 10% distortion. If the measured value is higher than 10% Fail will be incremented by 1. If Fail equals to zero PASS will be displayed, if Fail is greater than zero FAIL will be displayed.

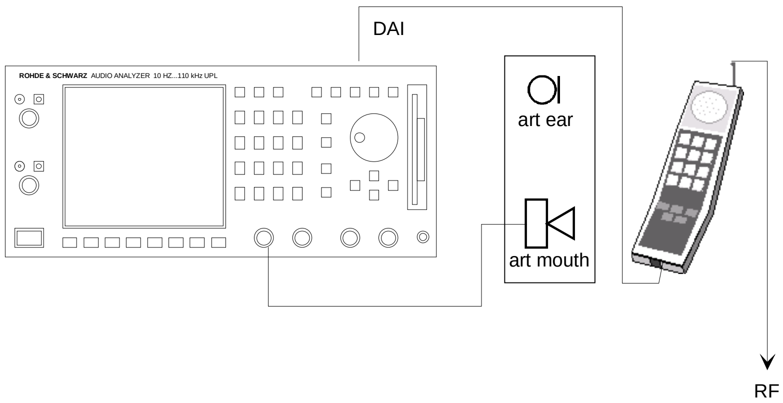
30.9 Out-of-band signals

30.9.1 Sending

30.9.1.1 Definition and applicability

The discrimination against out-of-band input signals in the sending direction is a requirement on the in-band image frequencies created by any out-of-band input signals. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.9.1.4 Method of test

30.9.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51. The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.9.1.4.2 Procedure

a) A pure tone with a sound pressure of -4,7 dBPa is applied at the mouth reference point as described in ITU-T P.64 using an artificial mouth conforming to ITU-T P 51.

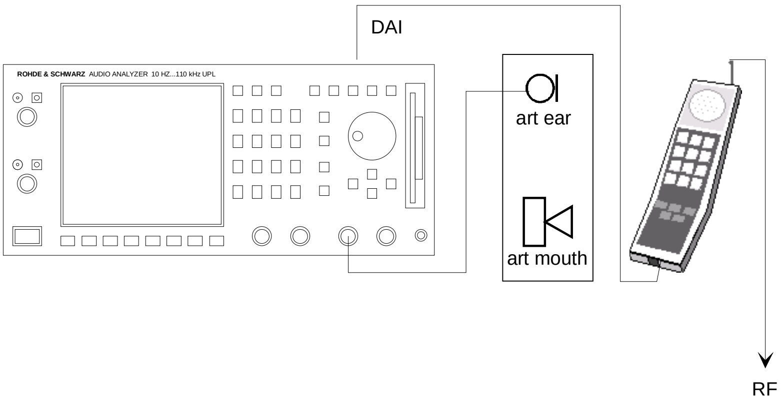
b) For input signals at frequencies of 4,65, 5, 6, 6,5, 7, and 7,5 kHz, the level represented by the PCM bit stream at the DAI (Pin 23) of any image frequency is measured.

30.9.2 Receiving

30.9.2.1 Definition and applicability

The discrimination against out-of-band signals in the receiving direction is a requirement on the out-of-band signals generated in the artificial ear from in-band input signals. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.9.2.4 Method of test

30.9.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51. The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.9.2.4.2 Procedure

a) The SS sends over the DAI (pin 25) a PCM bit stream simulating a sine-wave signal with a level of 0 dBm0.

b) For input signals at the nominal frequencies 500, 1 000, 2 000, and 3 350 Hz (bearing in mind the restriction on sub-multiples of the sampling frequency) the level of any out-of-band signals at frequencies from 4.6 kHz up to 8 kHz is measured in the artificial ear.

Procedure:

The input signals are sent in sequence, the measurement is a FFT with max hold function switched in. So the residual displayed curve shows all signal components for any input signal. If the curve is below the limit line, no single out-of-band component has exceeded the limit.

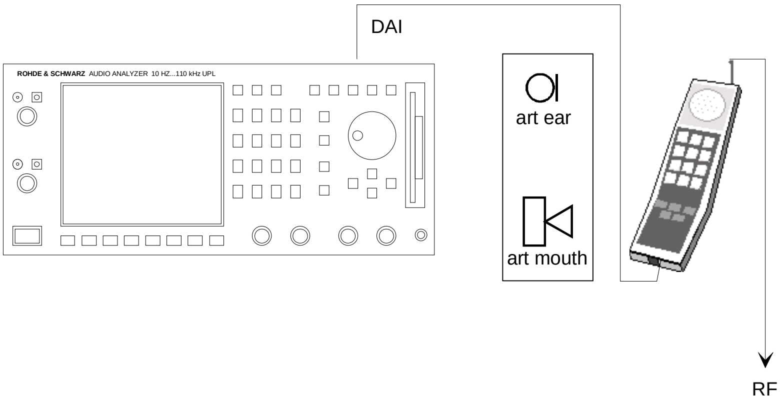
30.10 Idle channel noise

30.10.1 Sending

30.10.1.1 Definition and applicability

The idle channel noise in the sending direction is the equivalent noise level produced at the DAI, when the mouth reference point is in a quiet environment. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.10.1.4 Method of test

30.10.1.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51 in a quiet environment (ambient noise less than 30 dBA). The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.10.1.4.2 Procedure

The noise level represented by the PCM bit stream output at the DAI (pin 23) is measured with psophometric weighting according to ITU-T G.223, table 4.

NOTE: *The ambient noise criterion should be met if the ambient noise does not exceed NR20.*

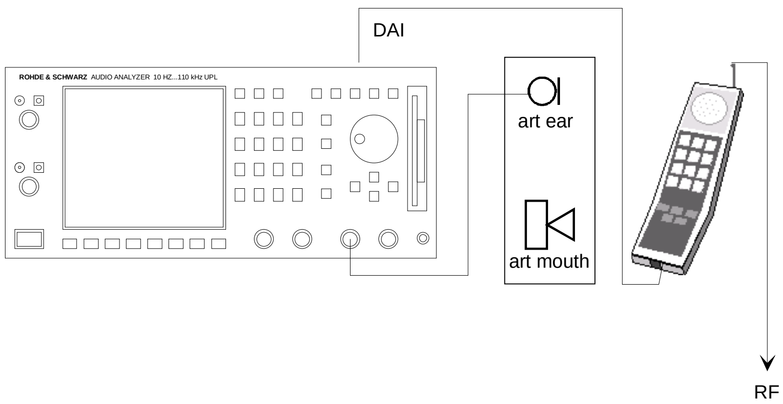
30.10 Idle channel noise

30.10.2 Receiving

30.10.2.1 Definition and applicability

The idle channel noise in the receiving direction is the acoustic sound pressure in the artificial ear when the digital input signal at the DAI, is the PCM coded value No 1. The requirements and this test apply to all types of GSM 900 and DCS 1 800 handset MS supporting speech.

Block diagram



30.10.2.4 Method of test

30.10.2.4.1 Initial conditions

The handset is mounted in the LRGP (see annex 1 of ITU-T P.76) and the earpiece is sealed to the knife-edge of the artificial ear conforming to ITU-T P.51. The DAI of the MS is connected to the SS and is set to the operating mode "Test of acoustic devices and A/D & D/A ".

30.10.2.4.2 Procedure

a) The SS sends a PCM bit stream coded with the value No 1 over the DAI (Pin 25) to the MS.

b) The level of the noise is measured in the artificial ear with any volume control set at the position at which the RLR is equal to the nominal value.

c) Where a volume control is provided, the level of the noise is measured in the artificial ear with the volume control set to maximum.

The sound pressure level is measured in the artificial ear using A-weighting filter according to IEC 179

30.11 Ambient Noise Rejection

This testcase 30.11 is valid up to Rel-99. From Rel-4 it is replaced by 30.19 (according to 3GPP TS 26.132) which is available with Upgrade UPL-U81.

Annex B: BASIC control program for test cases

The control program MAIN.BAS and MAIN32.BAS is supplied with UPL16. With this program all the test cases can be executed using Radiocommunication Tester CMD 52, 55, 65 or CMU200. The optional Real-Time Speech Coder/Decoder CMD-B52 or CMU-B52, is required for the test cases Echo Loss and Stability Margin.

The MAIN.BAS program requires the use of ear type 1, whereas the MAIN32.BAS program requires ear type 3.2 low leakage.

A standard PC keyboard has to be connected to UPL16 for starting and loading the program.

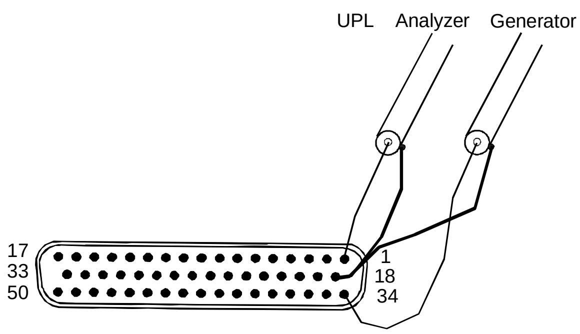
The program itself and the subprograms required for running the tests are stored in the directory C:\GSM of UPL. If a different working directory was used in the UPL before, directory C:\GSM has to be selected prior to program loading. The path can be changed in the following ways:

- in the manual mode with the "Working Dir" command in the FILE panel
- by calling up a setup required for the test cases
- from the universal sequence controller with the BASIC command line
  UPL OUT "MMEM:CDIR 'C:\GSM'"
- under BASIC via the SHELL command by entering CD C:\GSM and EXIT
- at DOS by entering CD C:\GSM

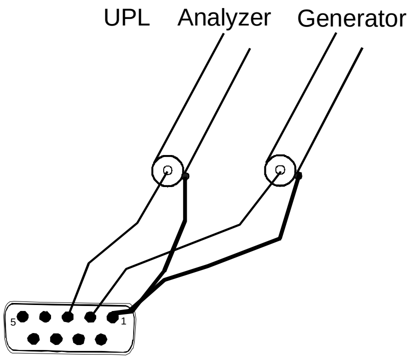
Starting the program:

Call up the programming interface of UPL16 with function key F3 on the PC keyboard. If you want to use artificial ear type 1, load the MAIN program by entering LOAD"**MAIN**" <RETURN> and start it with RUN <RETURN>. The program **MAIN32** is used for artificial ear type 3.2. If a DOS error message is displayed during program loading, it can be assumed that the working directory is not correct. In this case change the directory with command UPL OUT "MMEM:CDIR 'C:\GSM <RETURN> as described above.

To ensure proper function of the program all connectors must be correctly fitted and all instruments, microphone feeds, etc, must be switched on. A suitable cable is required for the test cases Echo Loss and Stability Margin which has to be connected to the multifunction connector of CMD or CMU.



Assignment of 50-contact multipoint connector at CMD front panel



Assignment of 9-contact speech connector at CMU front panel

Measurements:

The individual measurements are called up with the respective function keys on the UPL or the keyboard.

Prior to starting the measurements, a call has to be set up to the test mobile by the CMD or CMU and the speech mode "Handset" selected on. It is advisable to prepare a setup for the CMD or CMU that can then be used for the measurements.

Make sure that the test mobile is in the CALL state and that the DAI interface is properly initialized and connected to the DAI interface of UPL16. Each time a measurement is started, the presence of the DAI clock of the test mobile is checked. If the clock is not present, FAIL is indicated and an error message output.

The frequency response measurement results are graphically displayed after each test or numerically output together with a PASS or FAIL mark, depending on whether the test conditions have been met or not.

After completion of each measurement there is the choice to make a hardcopy, to store results or to continue the program.
Pressing the CONT key brings back the selection menu for the measurement.

When the TRC_FILE key is pressed, the displayed trace is saved in ASCII format in a file. This file has the name TRCxx.TRC, with xx representing a consecutive number (of max. 5 digits). This allows processing of measurement results with other programs. The TRC_FILE key has to function when the results are numerically displayed.

The screen content can be copied into a PCX file using the PCX_FILE key. This file has the fixed name PICxx.PCX, with xx representing a consecutive number (of max. 5 digits). Thus the measurement results can also be used in word processing programs, for instance. To allow also numeric values to be stored in a PCX file, the whole screen content without the softkey line is copied.

Since both the TRC and the PCX files are consecutively numbered, it is useful to copy the files of a measurement sequence, for instance, and to save them under a new name. In this case the original TRCxx.TRC and PICxx.PCX files can be cleared. Thus results can be identified more easily and a mix up between them avoided. (The files can be copied and renamed using common DOS commands.)

The screen content can be output to a printer by pressing the HARDCOPY key.
Printer type and desired settings are not selected by the program but the printer selected last and set in UPL manual mode will be chosen. For this reason the desired printer, scaling and format should be manually set once in the OPTION panel of UPL prior to the measurement.

Entering EXIT <RETURN> brings back BASIC without the program being cleared. The program can be restarted immediately by entering RUN.

Annex C

MOBILE PHONE TESTS

GSM RELEASE 99 with CMU

UPL16 UPGRADE

UPL-U81

1154.7900.02

Version 1.0

Contents

1 Overview

The Audio Analyzer UPL16 was developed for acoustic measurements on GSM mobiles with DAI (digital audio interface). The UPL16 has a DAI in line with GSM phase 2, 11.10 section 36.4 (now 3GPP TS 44.014 section 10).

Together with the Radiocommunication Tester CRTC, the UPL16 provides all acoustic tests on GSM phase 2 mobiles. The test cases required for type approval testing have been validated. They will still be available after the UPL-U81 upgrade has been installed for type approval tests using the UPL16 and the CRTC.

Using the UPL-U81, all acoustic tests in line with 3GPP TS 51.010 Release 99 are installed on the UPL16. These tests run on the Radio Communication Tester CMU200. All tests required for type approval testing are validated.

Acoustic Tests on GSM Mobile Phones with Option UPL-U81 (DAI)	**3GPP TS 51.010** **GSM Release 99** (DAI) Chapter
Sending sensitivity/frequency response	**30.1***
Sending loudness rating	**30.2***
Receiving sensitivity/frequency response	**30.3***
Receiving loudness rating	**30.4***
Sidetone masking rating	**30.5.1***
Listener sidetone rating	30.5.2
Echo loss	30.6.1 (release 4)
Stability margin	**30.6.2***
Sending distortion	**30.7.1***
Receiving distortion	30.7.2
Sidetone distortion	30.8
Out-of-band signals (sending / receiving)	30.9
Idle channel noise (sending / receiving)	30.10
Ambient noise rejection	30.11 (release 4)

*** validated testcases**

2 Preparation and Start of Application Software

Required Measuring Instruments and Accessories

The Audio Analyzer UPL16 with upgrade UPL-U81 and firmware 3.01 or higher is required for the measurements.

The GSM mobile phone under test is driven by the Digital Radio Communication Tester CMU200 via the RF interface. This tester simulates a base station so that a call can be set up. The Radio Communication Tester CMU200 must be equipped with the options CMU-B21 (versatile signalling unit), CMU-B52 (speech codec) and the appropriate software options for the GSM band used. The CMU200 must be equipped with firmware 3.0 or higher.

Acoustic devices such as an artificial mouth, artificial ear and other accessories are required for the measurements. The following equipment from Brüel & Kjaer or G.R.A.S. is normally used (also HATS can be used):

Device	Description	Type
Telephone test head	Device for fixing the DUT in the prescribed position	B&K 4602B
Ear simulator	Measuring microphone with adapters for connection to the ear piece of the DUT	B&K 4185 (type 1)
Wideband ear simulator	Measuring microphone with adapters for connection to the ear piece of the DUT	B&K 4195 (type 3.2)
Head and torso Simulator	Artificial head with adapter for the DUT with built-in artificial mouth and ear	B&K 4128D (type 3.3)
Artificial mouth	Special loudspeaker for simulation of the mouth	B&K 4227
Acoustic calibrator	Sound level calibrator for calibrating the measuring microphone	B&K 4231
Microphone power supply	Power supply and preamplifier for the measuring microphone	B&K 2690A0S2 or G.R.A.S. AA12

Note: *With the preamplifier set to 0 dB, the microphone power supply B&K 2690A0S2 produces too much noise for measuring idle noise and distortion. It is therefore advisable to set a gain of 20 dB. A low-noise power supply such as AA12 from G.R.A.S is preferable.*

A cable with a BNC connector and a special small or angled banana plug is required for connecting the artificial mouth, as the space between the mouth connector and the test rack (B&K 4602B) is too small for common banana plugs.

The transformer in the rear panel of the UPL16 is connected between the generator output 1 of the Audio Analyzer UPL16 and the connector of the artificial mouth. The transformer matches the impedance of the loudspeaker in the artificial mouth to that of the UPL16 generator output. Without this transformer, the available power is too low for driving the artificial mouth.

Alternatively, a power amplifier preferably with a gain of approx. 0 dB can be connected between generator output and mouth instead of the transformer. In this case, the gain set must be kept absolutely stable after calibration.

Together with the UPL-U81 upgrade, a cable with male (analyzer) and female (generator) XLR connector is supplied for connection to the "Speech" connector of the Radiommunication Tester CMU200.

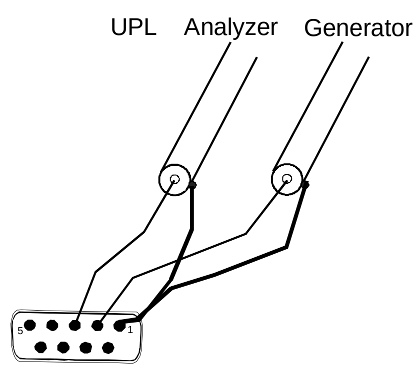


Fig. 1 Assignment of 9-contact speech connector on CMU front panel

An external PC keyboard must also be connected to the UPL16 (large DIN connector). A driver for country-specific keyboards can be defined in the C:\UPL\USERKEYB.BAT file (see the UPL manual, section 2.15.4).

The BASIC programs required for automatic sequence control are on the floppy supplied with the UPL-U81 upgrade option. The audio analyzer must meet the following firmware requirements:

- UPL16 firmware version 3.01 or higher
- UPL-U81 upgrade (Mobile Phone Tests GSM Release 99) installed
- UPL16 configured with 64 kbyte program memory and 32 kbyte data memory for automatic sequence control (using configuration tool UPLSET setting 3).

Installing the Software

The application software is installed with the aid of the U81INST.BAT installation program on the program floppy. The installation number of the UPL-U81 upgrade (Mobile Phone Test GSM Release 99) must be known.

Caution: *The software can only be installed on the specified Audio Analyzer UPL16 with matching serial number.*

➢ Quit the measurement software by pressing the SYSTEM key on the instrument or Ctrl + F9 on the keyboard.

➢ Insert floppy

➢ Select floppy disk drive (enter A:).

➢ **Call the installation program (enter U81INST).**
You are requested to enter the installation number of the UPL-U81 upgrade.

➢ **Enter the installation number supplied with the UPL-U81. If the number does not match the serial number of the UPL16, the installation is aborted.**

➢ **Return to the UPL program (enter C:\UPL).**

Test Setup

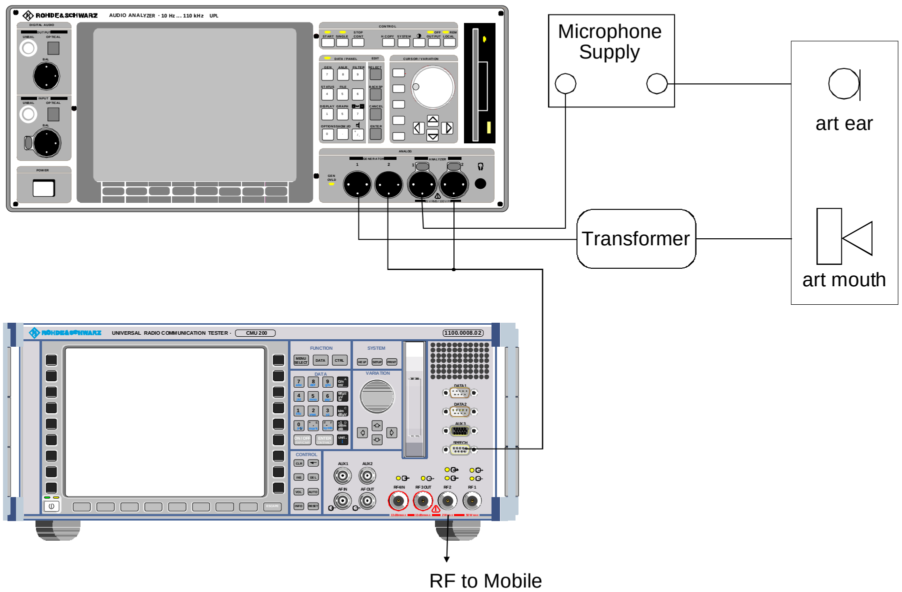


Fig. 2 Test setup and connection of external components

Starting the Application Software

The application program is executed by the universal sequence control. The Audio Analyzer UPL16 is switched to automatic sequence control using the F3 key (on the external keyboard).

The logging function is switched off; check that "logging off" is displayed at the bottom right of the screen; use the F2 key to enable/disable the logging function. With the logging function on, all commands entered in the manual mode would be appended to the program and so use up memory.

The application programs must be called from path C:\GSM in order to find all the required program routines and setups. The path can be changed in any of the following ways:

- in the manual mode with the "Working Dir" command in the FILE panel
- by calling one of the setups required for measurements on mobile phones
- in the automatic sequence control mode with the BASIC command
  UPL OUT "MMEM:CDIR ’GSM"
- under BASIC with the SHELL command by entering CD\GSM and pressing EXIT
- at DOS level by entering CD\GSM.

Program floppy contains the BASIC program R99_TST.BAS for measurements on GSM mobile phones. It is loaded and started by entering:

- LOAD"R99_TST"
- RUN

The softkeys displayed at the bottom of the screen in the automatic sequence control mode can be used instead.

Configuring the Application

"Default-Printer" is factory-set in the OPTION panel. This means that the printer configuration does not depend on the setup, but that the printer used last by the Audio Analyzer UPL16 remains configured. New settings need not therefore be made by the user. It is useful to select the desired printout with type, format, and scaling in manual mode before the program is started. All subsequent printouts triggered with the hardcopy key will then be printed with these settings.

IMPORTANT: *Correct execution of the software cannot be guaranteed if settings in the setup are changed.*

Setup Conversion for Firmware Updates

For an update of the UPL16 firmware, the setups may have to be converted. This is done automatically when the setup is loaded, but the conversion delays the loading. To avoid the delay, the setups can be converted before the application software is started:

- at DOS level by calling the UPL16 conversion program:

  DO_CONV \GSM

  This converts all setups in the GSM directory.

IMPORTANT: *Please note that a previous firmware version can no longer be used in the UPL16 after the conversion of setups.*

3 Operating Concept

Softkeys are displayed at the bottom of the screen for operation and test program selection. The softkey functions are also assigned to hardkeys on the external keyboard so that the keyboard can be used for selecting program routines.

After the program has been started, the title page

Measurement of
GSM Mobile Phones
with Audio Analyzer UPL16
according to 3GPP TS 51.010
release 99 via DAI

and the following softkey line are displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After F6 CONT has been pressed, the following request is displayed:

Selection of
Ear Type used

F5	F6	F7	F8	F9	F10	F11	F12
	TYPE 1	TYPE 3.2L	TYPE 3.2H	TYPE 3.3	TYPE 3.4		

After a type has been selected, the following request is displayed on the screen:

Please establish

call to Mobile

and set CMD

to Bit Stream Handset Low

To do so switch on the mobile phone. After successful registration, press the CONNECT MOBILE key on the CMU or dial a number on the mobile phone and press the transmit key. The selection of the bit stream setting is possible only in the "Call established" state.

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After F6 CONT has been pressed, the following message is displayed:

Measurement of
GSM Mobile Phones
with Audio Analyzer UPL16
according to 3GPP TS 51.010
release 99 via DAI

select Test to be performed

The measurements on the mobile phone under test can now be started since all required calibration values are stored in the UPL16.

Important:

When the test setup is installed for the first time, the artificial ear, the artificial mouth and the voice coder in the CMU have to be calibrated (see "Calibration Routines"). In this case, the message requesting a call setup to the mobile phone under test can be skipped with CONT.

To select the individual measurements, the softkeys F5 to F12 with abbreviations for the measurement names are displayed.

F5	F6	F7	F8	F9	F10	F11	F12
END	SND_FRQ	SLR	REC_FRQ	RLR	STMR	LSTR	→

A click on a key starts the test routine associated. Since there are more selection items than softkeys, the next set of softkey definitions is called with the F12 key.

F5	F6	F7	F8	F9	F10	F11	F12
←	ECHO	STAB_MAR	DIST_SND	DIST_REC	DIST_SDE	OUT_SND	→

F5	F6	F7	F8	F9	F10	F11	F12
←	OUT_REC	IDLE_SND	IDLE_REC	AMB_NOI			→

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

If F12 shows an arrow towards the right, press F12 to see the next set of softkey definitions. Press F5 showing an arrow towards the left to go back to the previous set. At the lowest level F5 shows END. After pressing F5 END the query "Do you really want to quit?" is displayed and the program can be quit. (see page 41)

4 Measurements

General

Special problems are encountered when measuring acoustic characteristics caused by the coder and decoder algorithms of mobile phones. Vocoders are used to attain the lowest possible data rate. In this case, not the actual voice signals but only the filter and fundamental parameters required for signal reconstruction are transmitted.

Purely sinusoidal tones normally used for the measurements cannot be transmitted with such a system. In type-approval tests, the coder and decoder are excluded from the measurement. A mobile phone required a digital audio interface (DAI) for the transmission of audio signals with linear PCM coding. The Audio Analyzer UPL16 is equipped with this DAI interface, so that direct transmission of the test signals to and from a mobile phone under test with DAI interface is possible.

Notes on Individual Measurements

The measurements to be performed are described below in the sequence in which they are carried out.

Perform all measurements in an anechoic test chamber with sufficient absorption against interfering sound. Since special distortion measurements and the measurement of idle noise set high demands on measurement conditions, the A-weighted noise in the test chamber must be below 30 dB(A).

Measurements are started by pressing the corresponding softkey or function key on the external keyboard. When the measurement is completed, the results are shown and the following softkey line is displayed.

F5	F6	F7	F8	F9	F10	F11	F12
	CONT		ABS_SENS	EXP_FILE	TRC_FILE	PCX_FILE	HARDCOPY

A return to the selection level is possible with CONT or the results can be printed or saved (see section 6, Processing of Measurement Results).

Sending Frequency Response

The sending frequency response is specified as the transmission ratio in dB of the digital signal value at the DAI to the input noise pressure at the artificial mouth.

The mobile phone under test is installed in the LRGP position (loudness rating guard ring position to ITU-T P.76), and the speaker is sealed to the artificial ear. Tones with a sound pressure of -4.7 dBPa are created with the artificial mouth at the MRP (mouth reference point), and the digital value is measured at the DAI and evaluated.

The sending frequency response must be within the tolerances specified according to table 30.1 of 3GPP TS 51.010. The absolute sensitivity is not yet taken into account.

Table 1 Tolerances specified according to table 30.1 of 3GPP TS 51.010

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-12	
200	0	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	0	

The offset of the measured frequency response to the upper or lower limit curve is calculated and then the whole trace is shifted by the mean value of the maximum and minimum offset. Then another limit check is performed. If the shifted curve is now within the limit lines, PASS is output, otherwise FAIL is displayed. The limit check is only performed at each measured frequency. It may happen that the trace slightly crosses a corner of the limit curve although there are no limit violations.

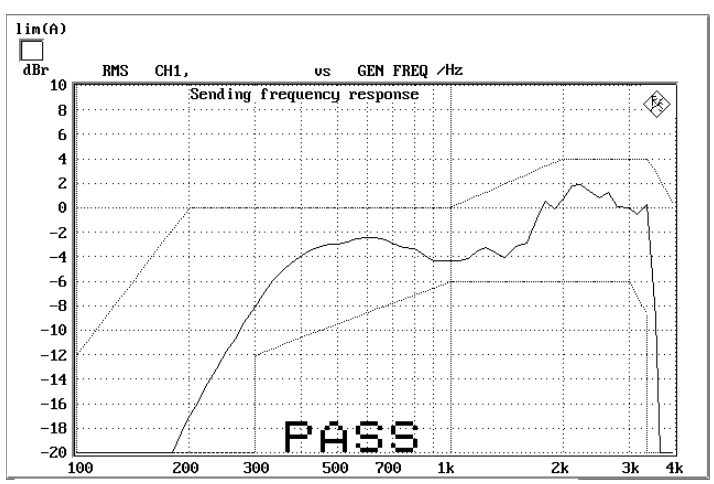


Fig. 3 Sending Frequency Response

Sending Loudness Rating

The sending loudness rating (SLR) takes into account the absolute loudness in the transmit direction and weights the tones in compliance with the normal sensitivity of the average human ear.

To this end the frequencies of bands 4 to 17 are evaluated according to table 1 of ITU-T P.79.

Table 2 Frequencies of bands 4 to 17 according to table 1 of ITU-T P.79

200	1000
250	1250
315	1600
400	2000
500	2500
630	3150
800	4000

The sensitivity at each frequency is defined as the ratio dBV/Pa referred to the digital FS = 3.14 dBm0, and the sending loudness rating is calculated according to formula 2.1 of ITU-T P.79.

According to 3GPP TS 51.010 the sending loudness rating should be between 5 dB and 11 dB, with lower dB values corresponding to greater loudness (5 dB = maximum loudness, 11 dB = minimum loudness). The measured SLR is checked for compliance with these limits. In addition, either PASS or FAIL is displayed.

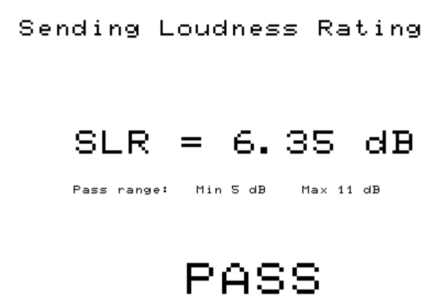


Fig. 4 Sending Loudness Rating SLR

Receiving Frequency Response

The receiving frequency response is specified as the transmission ratio in dB of the sound pressure in the artificial ear to the digital signal value at the DAI. The measured sound pressure is referred to the ear reference point (ERP). For ear type 1, the measuring microphone is directly applied to the ERP and a further correction is not required. For ear types 3.x, the measuring microphone is applied to the drum reference point (DRP) which is why any measured value has to be converted into the ERP by means of correction factors.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The signal is applied via the DAI, and the sound pressure in the artificial ear is measured and evaluated.

Table 3 Limit lines according to table 30.2 of 3GPP TS 51.010

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	2	

* Intermediate values are obtained when a straight line is drawn between the specified values, and a logarithmic frequency scale and a linear dB scale are used.

The offset of the measured frequency response to the upper or lower limit curve is calculated and the total curve shifted by the mean value of the maximum and minimum offset. Another limit check is then performed. If the shifted curve is within the limit lines, PASS is output, otherwise FAIL is displayed. The limit check is only performed at each measured frequency. It may happen that the trace slightly crosses a corner of the limit curve although there are no limit violations.

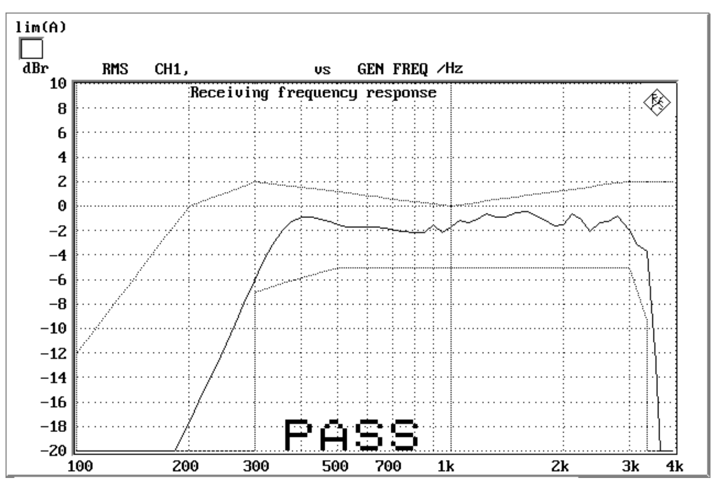


Fig. 5 Receiving Frequency Response

Receiving Loudness Rating

There are two routines since the permissible limit values for the loudness rating depend on the loudness set in the mobile phone under test. REC_NOM checks the compliance with the rated loudness setting and REC_MAX checks the maximum loudness.

The receiving loudness rating (RLR) takes into account the absolute loudness in the receive direction and weights the tones in compliance with the normal sensitivity of the average human ear.

To this end the frequencies (Hz) of bands 4 to 17 are evaluated according to table 1 of ITU-T P.79.

Table 4 Frequencies (Hz) of bands 4 to 17 to table 1 of ITU-T P.79

200	1000
250	1250
315	1600
400	2000
500	2500
630	3150
800	4000

The sensitivity at each frequency is specified as a ratio in dBPA/V referred to the rated internal signal level, and the receiving loudness rating is calculated according to formula 2.1 of ITU-T P.79.

The receiving loudness rating depends on the receiving loudness set on the mobile phone under test and, according to 3GPP TS 51.010, should be between -1 dB and +5 dB at a rated loudness setting, with lower dB values corresponding to a higher loudness.

The RLR may not fall below -13 dB when maximum loudness is set on the phone, i.e. the maximum receiving loudness may not exceed a certain value to avoid damage to the human ear.

The RLR measured is checked for compliance with these limits. In addition to the value measured, either PASS or FAIL is displayed.

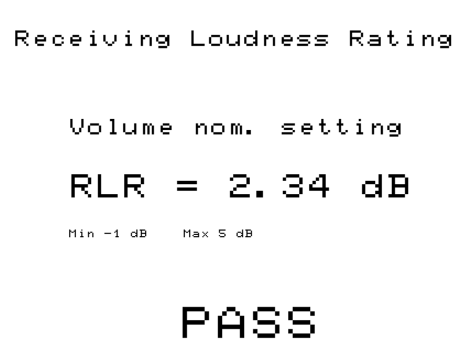


Fig. 6 Receiving Loudness Rating RLR

Sidetone Masking Rating (STMR)

The sidetone path is the deliberate output of part of the signal picked up by the microphone to the phone's receiver. This is to create a natural hearing impression for the person speaking on the phone as is encountered under normal conditions that involve an acoustic path between mouth and ear.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The STMR can only be measured with ear type 1 or ear type 3.2 Low Leakage.

The artificial mouth generates tones with a sound pressure of -4.7 dBPa at the MRP (mouth reference point), and the sound pressure is measured in the artificial ear.

The suppression of the sidetone path is determined at each frequency according to table 1 of ITU-T P.79, and the sidetone masking rating (STMR) is calculated according to formula 2.1 of ITU-T P.79 with the weighting factors of table 3 of ITU-T P.79 taken into account.

When the phone is set to the rated receiving loudness, the STMR should be between 8 dB and 18 dB according to 3GPP TS 51.010.

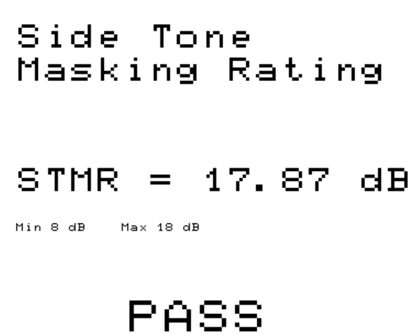


Fig. 7 Typical measurement of Side Tone Masking Rating

Listener Sidetone Rating (LSTR)

The listener sidetone rating defines the effect of interfering sound on the voice quality. The microphone of the telephone not only picks up the wanted voice but also any ambient noise.

To perform this measurement, a homogeneous noise field for simulating the ambient noise has to be generated. This field is identical to that for ambient noise rejection measurement.

This sound field must be generated by additional loudspeakers and noise generators. To obtain a sufficiently homogeneous sound field, several uncorrelated generators and loudspeakers are required. The use of 4 or 8 generators and loudspeakers is common practice. The noise sources have to generate pink noise (1/f). The permissible error in the relevant third-octave bands must be smaller than ±3 dB, with the frequency response of the loudspeakers used also being taken into account.

The LSTR measurement is divided into several measurements:

F5	F6	F7	F8	F9	F10	F11	F12
END	SND_FRQ	SLR	REC_FRQ	RLR	STMR	LSTR	→

After selection of LSTR, the following softkey line is displayed:

Ambient Noise Field Measurement

F5	F6	F7	F8	F9	F10	F11	F12
BACK		ADJUST		START			

The sound field must be set to the required sound pressure level of 70 dB(A) using ADJUST.

To measure the sound field, the reference microphone is used (as for mouth calibration). Fix the reference microphone to the mouth reference point (MRP) using a suitable support. For the sound field measurement, all other components such as artificial mouth or artificial ear must be removed.

After the ADJUST key has been pressed, a bargraph of the measured A-weighted sound pressure is displayed. It has to be set to a value of 70 ±1 dB. The tolerance limits are indicated by markers. Moreover, the bargraph colour changes from green to yellow if the value is outside the tolerance limits.

This adjustment of the sound field can practically be regarded as a calibration and has to be repeated only if something changes in the sound field generation. The long-term stability of the noise generators and loudspeakers must of course be sufficient. Otherwise, this measurement routine must be repeated as often as required.

The mobile phone under test is installed in the LRGP position (ITU-T P.76) and the speaker is sealed to the artificial ear.

The setup is installed in a chamber with a calibrated noise field. The energy distribution in this field is defined by 3GPP TS 51.010 and therefore known.

The energy of the sound pressure in the artificial ear is measured by means of third-octave analysis in the 14 bands with center frequencies from 200 Hz to 4000 Hz, and the suppression of the listener sidetone path is determined for each band from the known rated values of the sound field. The listener sidetone rating LSTR is then calculated with formula 8.4 of ITU-T P.79 by taking into account the weighting factors of tables 6 and 4 of ITU-T P.79.

The LSTR should not be less than 15 dB.

Fig. 8 Measurement of Listener Side Tone Rating

Echo Loss

The echo loss measurement to 3GPP TS 51.010 with artificial voice usually does not yield a PASS verdict. This problem was taken into account by the latest specifications and a new version of the echo loss measurement was created. This version is nominally validated only for GSM Release 4 or 3GPP mobiles, but contrary to the previous version provides accurate measurement results. The implementation of the echo loss measurement conforms thus not just to Release 99, but already to Release 4.

The echo loss is the attenuation between the voice coder input and the voice decoder output (gain of voice coder + decoder = 1). Normally the echo loss is caused by internal acoustic coupling between the telephone receiver and the microphone. Since the echo considerably reduces the sound transmission quality, it may not exceed a certain value.

The mobile phone under test is placed in the anechoic chamber at no specific position (former releases used the LRGP position!).

A modulated multitone signal to ITU-T P.501 is generated as a test signal and applied to the voice coder. First, the spectral energy distribution of the generated signal is measured in the third-octave bands from 200 Hz to 4 kHz. Then, the spectral distribution in the output signal of the voice decoder is measured. The echo loss is calculated from the differences of the individual bands according to ITU-T G.122. As an option, the mobile phone under test can be fed for approx. 10 s with the male and female version of artificial voice according to ITU-T P.50 prior to this measurement. This training sequence is to facilitate optimization for potential echo cancellers.The actual gain of the voice coder and decoder must also be considered in the result. This value is available in the CMU after calibration of the coder.

3GPP TS 51.010 specifies an echo loss of at least 46 dB which can be achieved by mobile phones using good echo cancellers. Since the microphone also picks up any side noise and treats it like an echo, it is essential that the test chamber is shielded against external noise.

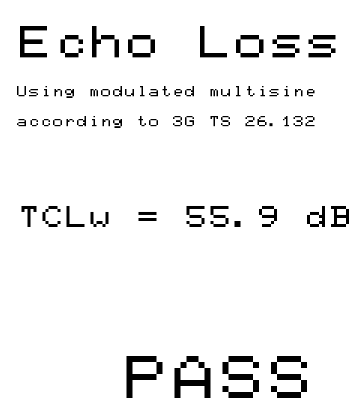


Fig. 9 Typical result of echo loss measurement

Stability Margin

The stability margin is measured to test the susceptibility of the phone to acoustic feedback and instability.

For the test, the telephone is placed on an even, hard board with the receiver and microphone pointing downwards.

A loop is closed in the UPL16 between the receiving and the voice channel and an overall gain of 6 dB is set. The gain of the coder is automatically taken into account (see also echo loss).

To activate the loop, a noise signal of -10 dBm0 in line with ITU-T O.131 is applied for 1 s and then switched off, with the loop remaining closed.

The test person has to listen whether resonances or oscillations are produced. If there are no oscillations, the minimum requirements to 3GPP TS 51.010 for a stability margin of 6 dB are complied with. This routine does not yield any measurement result!

Sending Distortion

The S/N ratio in the transmit path is measured as a function of the sound level.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The test signal is generated with the artificial mouth at the MRP (mouth reference point), and the SINAD value of the received signal is measured at the DAI.

The acoustic reference level (ARL) is defined as the sound pressure which creates a signal level of -10 dBm0 in the transmit channel. An automatic routine varies the sound pressure at the artificial mouth until the desired level is attained. This value is then used as a reference for determining the SINAD value versus level.

The SINAD value is measured at sound pressures between -35 dB and +10 dB relative to the acoustic reference level (ARL) and compared with the limit lines specified in table 30.3 of 3GPP TS 51.010.

Table 5 Limit lines specified in table 30.3 of 3GPP TS 51.010

dB relative to ARL	Level ratio
-35 dB	17.5 dB
-30 dB	22.5 dB
-20 dB	30.7 dB
-10 dB	33.3 dB
0 dB	33.7 dB
7 dB	31.7 dB
10 dB	25.5 dB

The measurement is performed up to a maximum sound pressure of 10 dBPa at the artificial mouth if the value 10 dB relative to ARL with 10 dBPa cannot be attained. The actual trace may therefore end at a lower pressure. This occurs for mobile phones under test which have a low sensitivity in the transmit direction.

If the measured trace is above the limit line, PASS is output, otherwise FAIL is displayed.

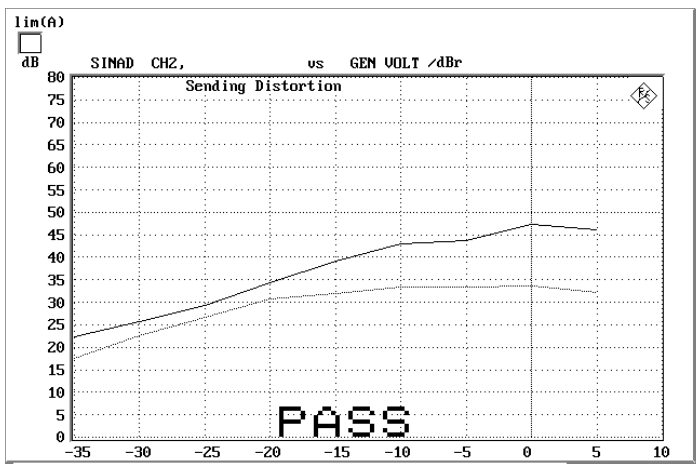


Fig. 10 Sending distortion measurement

Receiving Distortion

The S/N ratio in the receiving path is measured as a function of the sound signal level.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The test signal is applied to the mobile phone via the DAI, and the SINAD value of the sound pressure in the artificial ear is measured with psophometric weighting to ITU-T G.714.

The SINAD value of the sound pressure is measured at levels between -45 dBm0 and 0 dBm0 and compared with the limit lines given in table 8 of 3GPP TS 51.010.

Table 6 Limit lines given in table 30.4 of 3GPP TS 51.010

Level	Level ratio
-45 dBm0	17.5 dB
-40 dBm0	22.5 dB
-30 dBm0	30.5 dB
-20 dBm0	33.0 dB
-10 dBm0	33.5 dB
-3 dBm0	31.2 dB
0 dBm0	25.5 dB

The measurement is performed up to a maximum sound pressure of 10 dBPa in the artificial ear, so that the actual trace may end at a lower pressure.

If the measured trace is above the limit line, PASS is output, otherwise FAIL.

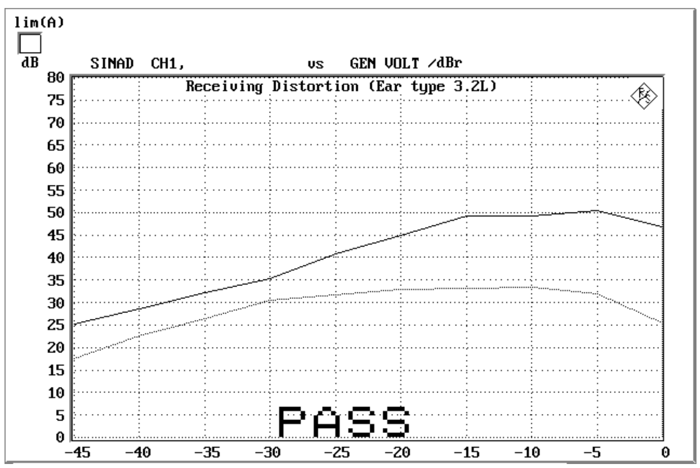


Fig. 11 Typical Result of Receiving Distortion Measurement

Sidetone Distortion

The cubic distortion of the sidetone signal coupled between microphone and telephone speaker in the mobile phone is measured.

The mobile phone under test is installed in the LRGP position (ITU-T P.76) and the speaker is sealed to the artificial ear.

The test signal is generated with the artificial mouth at the MRP (mouth reference point) and the sidetone signal distortion in the artificial ear is measured at the signal frequencies 315 Hz, 500 Hz and 1000 Hz.

The distortion of the sidetone signal must be less than 10 %.

```
Side Tone
Distortion

Distortion D3 @  315 Hz = 0.34 %
Distortion D3 @  500 Hz = 0.18 %
Distortion D3 @ 1000 Hz = 0.11 %

PASS
```

Fig. 12 Sidetone distortion measurement

Out of Band Sending

Signal components in the voice band up to 4 kHz are measured. They can be generated by acoustic triggering of the mobile phone with signals in the 4.6 kHz to 7.5 kHz range and may be due to inadequate filtering or unwanted mixer products.

The mobile phone under test is installed in the LRGP position (ITU-T P.76) and the speaker is sealed to the artificial ear.

The test signal is generated with the artificial mouth at the MRP (mouth reference point), and the received signal is band-filtered and measured at the DAI.

The products in the frequency range up to 4 kHz generated by out-of-band signals must be below the limit line to 3GPP TS 51.010, table 30.5.

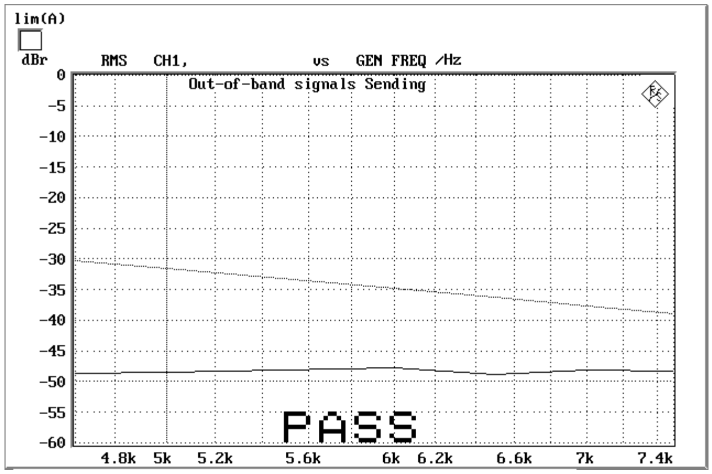


Fig. 13 Out-of-band Signals Sending Measurement

Out of Band Receiving

Signal components above 4 kHz are measured in the artificial ear which can be caused by triggering the mobile phone with signals in the range up to 4 kHz. These components may be due to inadequate filtering, unwanted mixer products or harmonic distortion in the receiver.

The mobile phone under test is installed in the LRGP (ITU-T P.76) and the speaker is sealed to the artificial ear.

The test signal is fed via the DAI, and the spectrum of the acoustic signal in the artificial ear in the frequency range above 4 kHz is measured.

The products generated in the frequency range above 4 kHz by means of the signals in the range up to 4 kHz must be below the limit line to 3GPP TS 51.010, table 30.6.

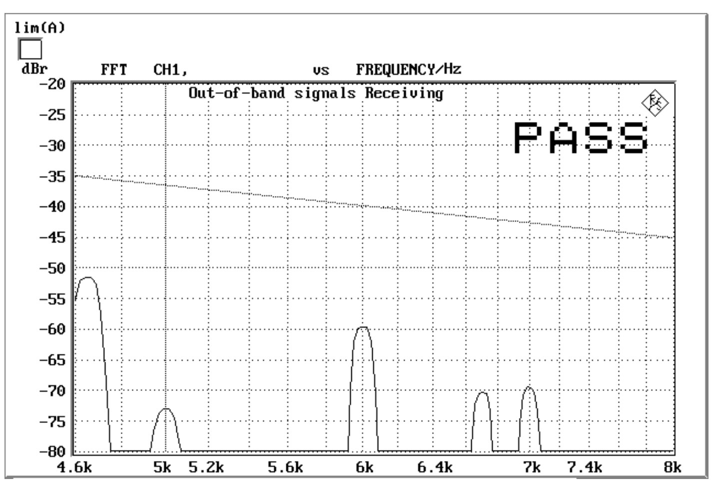


Fig. 14 Out of Band Receiving Measurement

Idle Channel Noise Sending

The idle noise signal at the DAI is measured with the telephone set up in a quiet environment (< 30 dB(A)).

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The digital signal at the DAI is measured, psophometrically weighted according to ITU-T G.223 and calculated at the internal level in dBm0p.

The idle noise level should not exceed -64 dBm0p.

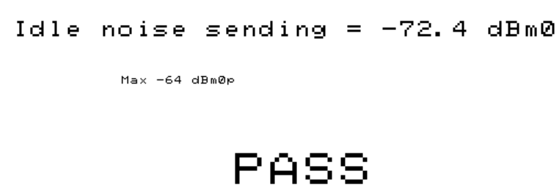


Fig. 15   Idle noise sending

Idle Channel Noise Receiving

The sound pressure in the artificial ear is measured with the phone set up in a quiet environment.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The sound pressure in the artificial ear is measured with A-weighting on.

With rated loudness set on the mobile phone, the sound pressure should not exceed -57 dBPa(A).

At maximum receiving loudness, the sound pressure should not exceed -54 dBPa(A).

This measurement makes high demands on the sound insulation of the test chamber and the S/N ratio of the measuring microphone including preamplifier in the artificial ear. A comparison measurement with the mobile phone under test switched off or without a DUT shows the measurement reserves of the test equipment. Due to the inherent noise of the Audio Analyzer UPL16, measurements can be made to about -80 dBPa(A) at 0 dB microphone gain, and even to lower values when a higher microphone gain is set.

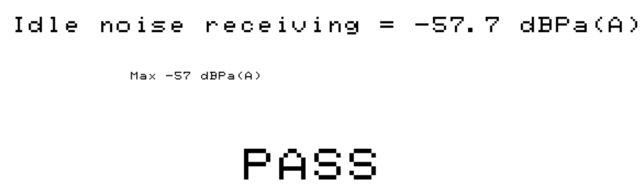


Fig. 16 Idle noise receiving

Ambient Noise Rejection

The ambient noise rejection measurement is not precisely defined for Release 99 in 3GPP TS 51.010. This problem was taken into account by the latest specifications and a new version of the ambient noise rejection measurement was created. This version is nominally validated only for GSM Release 4 or 3GPP mobiles, but unlike the older version provides accurate measurement results. The implementation of the ambient noise rejection measurement thus conforms not to Release 99, but already to Release 4.

Ambient noise rejection (ANR) describes the weighted ratio of voice signal transmission to ambient noise. An ANR value >0 dB means that voice as the useful signal is transmitted more loudly than any ambient noise. The minimum requirement to 3GPP TS 51.010 is ANR > -3 dB. A value >= 3 dB should at least be attained.

To perform this measurement, a homogeneous noise field for simulating the ambient noise has to be generated. The same noise field is also used to measure the LSTR.

This sound field must be generated by additional loudspeakers and noise generators. In order to obtain a sufficiently homogeneous sound field, several uncorrelated generators and loudspeakers are required. The use of 4 or 8 generators and loudspeakers is common practice. The noise sources have to generate pink noise (1/f). The permissible error in the relevant third-octave bands must be smaller than ±3 dB with the frequency response of the loudspeakers used also being taken into account.

The measurement of ambient noise rejection is divided into several single measurements.

F5	F6	F7	F8	F9	F10	F11	F12
←	OUT_REC	IDLE_SND	IDLE_REC	AMB_NOI			→

After selection of AMB_NOI the following softkey line is displayed:

Ambient noise field Measurement

F5	F6	F7	F8	F9	F10	F11	F12
← BACK		ADJUST		START			→

The sound field must be set to the required sound pressure level of 70 dB(A) using ADJUST.

To measure the sound field, the reference microphone is used (as for mouth calibration). Fix the reference microphone to the mouth reference point (MRP) using a suitable support. For the sound field measurement, all other components such as artificial mouth or artificial ear must be removed.

After the ADJUST key has been pressed, a bargraph of the measured A-weighted sound pressure is displayed. It has to be set to a value of 70 ±1 dB. The tolerance limits are indicated by markers. Moreover, the bargraph colour changes from green to yellow if the value is outside the tolerance limits.

After the CONT key has been pressed, the spectral distribution is measured automatically with an automatic check for compliance with the stipulated absolute level and the pink distribution. If a 1/f-weighted noise density (pink noise) is precisely complied with, a third-octave analysis yields identical amplitudes in each third-octave band. The tolerance of ±3 dB is complied with if the difference between the largest and smallest band measured is less than 6 dB. If this difference is larger, a warning will be output on the screen. A warning will also be output on the screen if the absolute sound pressure is outside the permissible tolerance.

This adjustment and testing of the sound field can practically be regarded as a calibration and has to be repeated only if something changes in the sound field generation. The long-term stability of the noise generators and loudspeakers must of course be sufficient. Otherwise, this measurement routine must be repeated as often as required.

For the actual measurement of ANR, the artificial mouth and the artificial ear must be installed again. The MRP must be installed at the same position as the reference microphone before.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker is sealed to the artificial ear.

Set up a call to the CMU and set the bit stream to "Handset Low".

Press the START key to start the measurement of the room noise sensitivity.

After completion of the measurement, the request to switch off the noise field will be displayed on the screen. If this is confirmed, the speech sending sensitivity will be measured automatically and the ANR value calculated afterwards.

The ANR values must exceed -3 dB.

Ambient Noise Rejection

ANR = 3.24 dB

PASS

Fig. 17 Typical result of ambient noise rejection measurement

5 Calibration Routines

All calibration data is stored on the UPL16 hard disk and is automatically available at every restart. The calibration values for the different ear types are stored independently from each other and other options. The calibration values for the artificial mouth are identical to the values for the older MAIN and MAIN32 programs.

Calibration of Artificial Ear

Before a mobile phone can be tested, the absolute sensitivity of the microphone in the artificial ear must determined using a sound level calibrator such as the Brüel & Kjaer 4231 with a sound pressure level of 94 dBSPL or a sound pressure of 1 Pa at 1 kHz.

Note:

The calibration values of the different ear types are stored separately. So, a calibration need not be performed after a change of the ear type if the physically identical ear has been calibrated before.

Calibration of Ear Type 1

➢ Switch off the microphone power supply.

Note: *The 200 V polarization voltage of the microphone may cause a slight electric shock. The current is harmless but the microphone preamplifier may be damaged*

➢ Remove the microphone from the artificial ear.

➢ Screw back the microphone capsule and switch on the operating voltage.

➢ Insert the microphone fully into the adapter of the sound level calibrator and switch on the calibrator.

Note: *After inserting the microphone wait at least 10 s to allow for static pressure compensation.*

➢ Select the CALIBRATION level using the F12 key.

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

➢ Call the calibration routines using the EAR key.

EAR TYPE1 **EAR TYPE3.2L** **EAR TYPE3.2H** **TYPE 3.3** **TYPE 3.4**

F5	F6	F7	F8	F9	F10	F11	F12
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34

- Call the test routine using the EAR_T1 key.

  The output voltage of the microphone is measured and the sensitivity displayed with reference to 1 Pa. With 20 dB preamplification of the microphone (recommended value), the sensitivity displayed must be about 10 times the value in the calibration certificate of the microphone capsule (typical value for microphone capsule 4134 and artificial ear 4185 is approx. 12 mV/Pa, display = 120 mV/Pa). If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched off microphone power supply or a disabled calibrator. In this case, the program requests the test to be repeated. After switching on the microphone power supply wait approx. 20 seconds before restarting the measurement with RUN.

  The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with the artificial ear type 1.

Calibration of Ear Type 3.2 Low Leakage

- Connect the noise level calibrator tightly to the artificial ear using the adapter DP0939 and switch on the calibrator.

- Select the CALIBRATION level using the F12 key.

---- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the EAR key.

EAR TYPE1 EAR TYPE3.2L EAR TYPE3.2H TYPE 3.3 TYPE 3.4

F5	F6	F7	F8	F9	F10	F11	F12
➢ BACK	➢ EAR_T1	➢ EAR_T32L	➢ T32L_DAT	➢ EAR_T32H	➢ T32H_DAT	➢ EAR_T33	➢ EAR_T34

- Call the test routine using the EAR_T32L key.

  The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched off microphone power supply or a disabled calibrator. In this case, the program requests the test to be repeated. After switching on the microphone power supply wait approx. 20 s before restarting the measurement with RUN.

  The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.2L.

Reading the Calibration Data of the Artificial Ear of Type 3.2L:

The frequency response of the artificial ear of type 3.2L is supplied on a floppy together with the artificial ear. The data is used for transforming the measurement values from the drum reference point to the ear reference point.

➢ Insert the floppy supplied with the ear into the UPL16 drive.

➢ Call the routine using the T32L_DAT key**.**

The OES_LL.ADA calibration file is automatically searched for and read. The modified data is stored on the UPL16 hard disk. This procedure need only be repeated after a change of the calibration data, e.g. after a recalibration of the ear by the manufacturer or when a physically different ear of the same type is used.

If the file required is not found on the floppy, the routine requests the user to insert the calibration floppy.

Calibration of Ear Type 3.2 High Leakage

➢ Connect the sound level calibrator tightly to the artificial ear using the adapter DP0939 and switch the calibrator on.

➢ **Select the CALIBRATION level using the F12 key.**

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

➢ Call the calibration routines using the EAR key.

EAR TYPE1 **EAR TYPE3.2L** **EAR TYPE3.2H** **TYPE 3.3** **TYPE 3.4**

F5	F6	F7	F8	F9	F10	F11	F12
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34

➢ Call the test routine using the EAR_T32H key.

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests the test to be repeated. After switching on the microphone power supply wait approx. 20 s before restarting the measurement with RUN.

The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.2H.

Reading the Calibration Data of the Artificial Ear of Type 3.2H:

The frequency response of the artificial ear of type 3.2H is supplied on a floppy together with the artificial ear. The data is used for transforming the measurement values from the drum reference point to the ear reference point.

- Insert the floppy supplied with the ear into the UPL16 drive.

- Call the routine using the T32H_DAT key.

The OES_HL.ADA calibration file is automatically searched for and read. The modified data is stored on the UPL16 hard disk. This procedure need only be repeated after a change of the calibration data, e.g. after a recalibration of the ear by the manufacturer or when a physically different ear of the same type is used.

If the file required is not found on the floppy, the routine requests the user to insert the calibration floppy.

Calibration of Ear Type 3.3

- Connect the sound level calibrator tightly to the artificial ear using the adapter UA-1546 and switch the calibrator on.

- **Select the CALIBRATION level using the F12 key.**

---CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the EAR key.

EAR TYPE1 EAR TYPE3.2L EAR TYPE3.2H TYPE 3.3 TYPE 3.4

F5	F6	F7	F8	F9	F10	F11	F12
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34

- Call the test routine using the EAR_T33 key.

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched off microphone power supply or a disabled calibrator. In this case, the program requests the test to be repeated. After switching on the microphone power supply wait approx. 20 s before restarting the measurement with RUN.

The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.3. The standard calibration data to ITU-T P57 are used automatically for ear type 3.3.

Calibration of Ear Type 3.4

- ➢ Remove the pinna and the ear canal simulator, connect the sound level calibrator tightly to the artificial ear using the short steel adapter and switch the calibrator on.

- ➢ **Select the CALIBRATION level using the F12 key.**

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- ➢ Call the calibration routines using the EAR key.

EAR TYPE1 EAR TYPE3.2L EAR TYPE3.2H TYPE 3.3 TYPE 3.4

F5	F6	F7	F8	F9	F10	F11	F12
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34

- ➢ Call the test routine using the EAR_T34 key.

  The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests the test to be repeated. After switching on the microphone power supply wait approx. 20 s before restarting the measurement with RUN.

  The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.4.

  The standard calibration data to ITU-T P57 are used automatically for ear type 3.4.

Calibration of Artificial Mouth

The calibration of the artificial mouth does not depend on the ear type used. A recalibration is therefore not required when the ear type is changed.

Before a mobile phone can be tested, the absolute sensitivity and frequency response of the artificial mouth have to be measured and corrected with the aid of a previously calibrated pressure-field measuring microphone. The measuring microphone removed from artificial ear type 1 can be used for this purpose or an additional microphone capsule is screwed to the microphone preamplifier. The measuring microphone is used as a reference for determining the frequency response of the mouth. The frequency response of the microphone can be ignored in the test frequency range (100 Hz to 8 kHz) (see also calibration certificate of microphone capsule).

Since interfering sound falsifies the corrections, the artificial mouth must be calibrated in a sound-proof test chamber.

First of all, the measuring microphone has to be calibrated.

- Insert the measuring microphone fully into the adapter of the sound level calibrator and switch on the calibrator.

Note: ***After inserting the microphone into the calibrator wait at least 10 s to allow for static pressure equalization.***

- Select the CALIBRATION level using the F12 key.

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the MOUTH key.

MOUTH CALIBRATION

F5	F6	F7	F8	F9	F10	F11	F12
BACK	REF_MIC	CAL_MOU					

- Call the test routine using the REF_MIC key.

  The output voltage of the microphone is measured and the sensitivity displayed with reference to 1 Pa. With 20 dB preamplification of the microphone (recommended value), the sensitivity displayed must be about 10 times the value in the calibration certificate of the microphone capsule (typical value for microphone capsule 4134 and artificial ear 4185 is approx. 12 mV/Pa, display = 120 mV/PA). If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched off microphone power supply or a disabled calibrator. In this case, the program requests the test to be

repeated. After switching on the microphone power supply wait approx. 20 s before restarting the measurement with RUN.
Fit the microphone at right angles to the mouth at the mouth reference point (MRP) using the gauge supplied with the mouth (positioning at right angles is necessary because microphone capsule 4134, e.g. of ear 4185, is pressure-calibrated).

- Select the CALIBRATION level using the F12 key.

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the MOUTH key.

MOUTH CALIBRATION

F5	F6	F7	F8	F9	F10	F11	F12
BACK	REF_MIC	CAL_MOU					

- Call the test routine using the CAL_MOU key.

The sound pressure generated at the MRP is set to exactly -4.7 dBPa in an automatic measurement routine at 1 kHz. The generator voltage required is stored in a nonvolatile memory and used as a reference for all subsequent settings with the artificial mouth. If the sound pressure cannot be adjusted to -4.7 dBPa, an error message is displayed with a request to check the connection of the artificial mouth and to repeat the measurement. A possible error source is that the transformer supplied is not connected between the generator and the artificial mouth.

The uncorrected frequency response of the artificial mouth is measured and displayed. Next, the frequency response is measured with the inverse frequency response correction automatically selected in the generator (equalization). Residual errors caused by non-linearities of the speaker in the mouth are measured and taken into account in the final equalization file as fine correction.

To verify the results, the absolute sound pressure versus frequency is measured at a sound pressure of -4.7 dBPa (reference value for most of the measurements). The absolute sound pressure at each frequency must be within a tolerance band of -4.7 dBPa ±0.2 dB. Correct calibration without interfering sound yields an almost straight line in the middle between the two limit lines.

Calibration of CMU Voice Coder

The calibration of the voice coder and decoder is required for the measurement routines performed via the voice path, such as stability margin, echo loss and ambient noise rejection. Calibration has to be performed only once and must be repeated only if the CMU used is replaced.

Auxiliary settings required for calibration can be found in the CMU under bit stream (firmware version 3.0 or higher). Since this menu is accessible only for active call, a call to a mobile phone has to be established first.

- Select the CALIBRATION level with the F12 key.

---CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the CMU-COD key.

The following information is displayed:

Calibration of

Coder – Decoder Path

in Radiocomm Tester CMU

Please establish call to mobile

and set Bit Stream to ‘Decoder Cal‘

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

Establish call to mobile phone, set bit stream on CMU to "Decoder Cal" and then press the CONT key.

The actual voltage at the decoder output of the CMU is now measured for a digital full-scale signal and the required correction value is calculated and saved in the UPL16. The following request is then displayed:

Calibration of

Coder – Decoder Path

in Radiocomm Tester CMU

now set Bit Stream to ‘Encoder Cal‘

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After CONT has been pressed, the input sensitivity of the voice coder is measured and the input voltage required for digital full scale is measured at the voice coder and saved in the UPL16. At the same time, the loop gain from the voice coder input to the voice coder output is calculated and saved.

6 Processing of Measurement Results

Printing, Storing and Displaying of Measurement Results

All results, the log file as well as all traces and pictures saved by keystroke are stored in the C:\GSM\RESULTS directory. This path is defined in the R99_TST.BAS program in the Path$ variable and can be modified, if required.

The result of each measurement is graphically or numerically displayed on the screen and, if applicable, a PASS or FAIL verdict is output. All individual numeric results such as loudness rating are automatically appended to a RES_99.LOG result file.

The following softkeys are displayed.

F5	F6	F7	F8	F9	F10	F11	F12
	CONT		ABS-SENS	EXP-FILE	TRC-FILE	PCX-FILE	HARDCOPY

The items ABS-SENS, EXP-File and TRC-FILE are displayed only if associated values are available for storage. The key name will be deleted after storage. The user can thus see any time whether the file has already been saved.

Pressing the CONT key brings back the selection menu for the measurements.

When the ABS-SENS key is pressed, the absolute measured values are saved in export format in a file. This file has the name ABSxx.EXP, xx representing a consecutive number (of max. 5 digits). It can directly be processed by spreadsheet programs such as EXCEL. This item is displayed only after a frequency response measurement.

When the EXP-FILE key is pressed, the displayed trace is saved in export format in a file. This file has the name EXPxx.EXP, xx representing a consecutive number (of max. 5 digits). This file can directly be processed by spreadsheet programs such as EXCEL.

When the TRC-FILE key is pressed, the displayed trace is saved in ASCII format in a file. This file has the name TRCxx.TRC, xx representing a consecutive number (of max. 5 digits). This allows processing of measurement results with other programs.

When the PCX-FILE key is pressed, the screen content is copied into a PCX file. This file has the name PICxx.PCX, xx representing a consecutive number (of max. 5 digits). Thus the measurement results can also be used in word processing programs, for example. To allow also numeric values to be stored as a PCX file, the whole screen content without the softkey line is copied.

Since the TRC, EXP, ABS and PCX files are consecutively numbered, it is useful to copy the files of a measurement sequence, for example, and to save them under a new name. In this case, the original files can be deleted. Thus results can be identified more easily and a mix up between them avoided.

To this end a DOS shell can be called after termination of the test program (e.g. with key F5) by entering the command SHELL <RETURN>. The files can be copied and saved under another name (as standard in the C:\GSM\RESULTS directory) using common DOS commands. Entering EXIT <RETURN> brings back BASIC without the program being deleted. The program can be restarted immediately by entering RUN.

The screen content can be output to a printer by pressing the HARDCOPY key.

Printer type and desired settings are not selected by the program but the printer selected last and set in the UPL16 manual mode will be chosen. For this reason, the printer, scaling and format must be manually set once in the OPTION panel of the UPL16 prior to the measurement. It is recommended to select a LOW or MEDIUM resolution and integer scale factors for the printer output. If fractional scale factors (especially values <1) are used, the pixel values are interpolated and the print quality is reduced.

It may be useful to first print a test copy to check the print quality. Contrary to manual operation, no COMMENT line is printed in this case and the program automatically sends a FORM FEED after each print to throw out the hardcopy.

All the purely numerical values are automatically added to the RES_99.LOGRES_99.LOG result file after each measurement. Thus all numeric measurement results can be called again and evaluated after a measurement sequence has been performed.

As with TRC and PCX files, it may be useful to copy the RES_99.LOG file after a measurement sequence and to save it under a new name. Afterwards the RES_99.LOG file can be deleted. Thus results can be identified more easily and a mix up between them avoided. To this end, a DOS shell can be called after termination of the test program (e.g. with key F5) by entering the command SHELL <RETURN>. The file RES_99.LOG can be copied and saved under another name using common DOS commands.

Entering EXIT <RETURN> brings back BASIC without the program being deleted. The program can be immediately restarted with RUN.

Old result files of a measurement sequence may be deleted by means of the DELETE EXP-FILES menu item in the R99_TST program.

➢ Select the CALIBRATION level using the F12 key.

--- CALIBRATION --- **EXP-FILES**

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

➢ Press the DELETE key.

A query is displayed whether all result files (also RES_99.LOG) are really to be deleted. If the user confirms, all result files are stored under *.OLD, i.e. they are not deleted right away. In a second delete procedure these backup copies are overwritten.

7 Terminating the Application

As long as the arrow → is displayed below the F12 key, another set of softkeys can be called using F12. With F5 the user can return to the previous set of softkeys, as long as the arrow ← is displayed below the key. If F5 displays END, there is no previous set.

F5	F6	F7	F8	F9	F10	F11	F12
END	SEND	REC_NOM	REC_MAX	LSTR	ECHO	STAB-MRG	→

If END is selected by pressing the F5 key, the following query is displayed:

➢ "Do you really want to quit?
<Y><N>“

Upon confirmation with Y, the program is aborted but not deleted. The softkey line for BASIC is automatically restored.

The software can be terminated any time under BASIC with the key combination CTRL BREAK. The program can be continued with CONT and restarted with RUN.