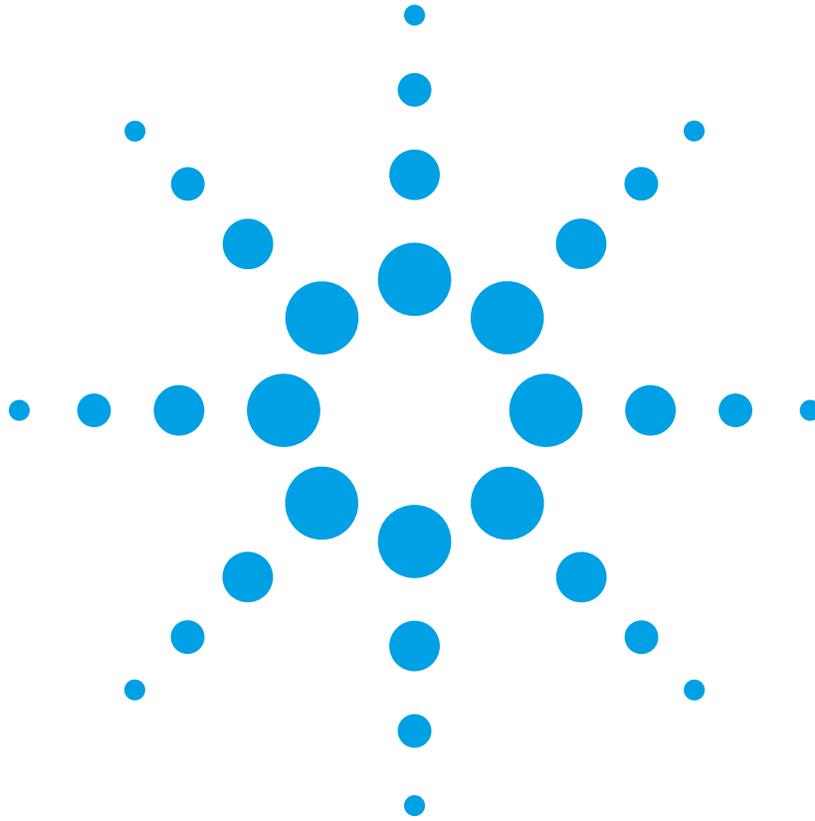# Understanding *Bluetooth*®
Audio Test with the N4010A
Wireless Connectivity Test Set

Application Note

Agilent Technologies

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1. Introduction

The use of *Bluetooth* technology for audio applications has grown over recent years. One of the major reasons is that *Bluetooth* has been readily adopted into mobile phones. With module manufacturing costs falling for both handsets and headsets, some manufacturers choose to include *Bluetooth* modules as standard. Similarly, phone retailers now increasingly offer a headset to complement the phone as a matter of course. Consumers now see *Bluetooth* technology as a cost-competitive alternative to wired headphones, headsets, and speakers. The technology is also better understood by consumers and the benefits are more readily recognized in a variety of phone applications. A specific example of mass consumer adoption is use in automobiles. This has been spurred by laws to ensure that drivers use hands-free devices for safety at the wheel.

Applications also extend beyond mobile phones. Wireless audio is enabled in a range of usage scenarios. Portable music players provide music on the move. By removing the need for a cable between the player and the earphones, the user can use the player more freely—in the gym, for example. With personal computers (PCs) now playing a greater role in the high tech digital home, *Bluetooth* has a variety of uses, such as wireless printing, passing photographs between devices, and controlling toys, to a name a just a few applications.

With the growth in *Bluetooth* audio, audio test has become a common manufacturing test requirement in *Bluetooth* test plans. These tests are necessary to check the audio quality of a device and the functionality of device components. Such tests are often carried out alongside radio frequency (RF) parametric tests. As a result, one box testers such as the Agilent N4010A Wireless Connectivity Test Set now incorporate audio functionality to complement existing parametric coverage.

Test setups for *Bluetooth* audio can vary depending upon many variables: the capability of the device, the test equipment, and test requirements. This application note shows how the N4010A can be used in a variety of different test scenarios to test *Bluetooth* audio. The flexibility of the tester is explained by introducing the different ways to route audio between the tester and device. This includes the use of the N4010A's internal audio generator and analyzer. Where appropriate, front panel screenshots and remote commands are provided to guide the user depending upon the audio routing chosen in the N4010A.

2. Introduction to Audio Measurements

2.1 The physical link

There are two types of physical links. A synchronous connection-oriented (SCO) link is used primarily used for audio. A SCO link is setup by the link manager and provides dedicated time slots for fixing the arrival time for packets. This makes it suitable for audio use. A device can support up to three audio channels and quality is comparable with that of GSM cellular technology. An asynchronous connectionless link (ACL) is used for data.

2.2 Codecs

Standard *Bluetooth* audio specifies three different codecs. The choice is either:

- a 64 kbps log pulse code modulation (PCM) format (A-law)

- a 64 kbps log PCM format (µ-law), or

- a 64 kbps continuous variable slope delta (CVSD)

The goal is to compress the data as much as possible while maintaining quality. This is a trade-off between bandwidth and signal to noise ratio, and depends on factors such as quantization levels, quantization characteristics, and the characteristics of the signal.

Log PCM is specified by the International Telecommunications Union (ITU-T) standard G.711. This standard defines the audio companding to be implemented to represent eight-bit compressed PCM samples for signals of voice frequencies. The sampling rate is 8 kHz and the encoding law uses eight binary digits per sample. Audio companding works on the assumption that in the case of voice signals, the statistical distribution of a person's speech signal amplitude can be modeled. The actual shape of the curve depends on geographic region—North America's µ-law or Europe's A-law. The A- and µ-laws are implemented as piecewise linear curves. The µ-law tends to give a slightly improved signal to quantization noise ratio when compared with the A-law, but it has a slightly smaller dynamic range.

CVSD is a more complex technique than log coding. The CVSD modulation is a nonlinear, sampled data, feedback system that accepts a band-limited analog signal and encodes it into binary form for transmission through a digital channel. It works by assuming a correlation between closely spaced samples of a signal and transmitting information about the change between samples instead of sending the sample values themselves. This technique is often referred to as differential PCM (DPCM) and the reduced number of bits per PCM codeword saves on bandwidth. Additionally, further digital compression can be achieved by adaptive DPCM (ADPCM), allowing for continuous adaptation of predictor coefficients to adjust to changing signal characteristics. If the quantizer of a DPCM system is set at two levels only, then the resulting scheme is called delta modulation (DM). With DM there is a compromise between optimization for quantization noise and slope overload. The requirement for acceptable quantization noise might be to reduce the step size as small as possible, while the requirement to reduce the effects of slope overload is to have the step size larger.

CVSD in *Bluetooth* is a delta modulation with variable step size, i.e. ADPCM with delta modulation as described above. It encodes a one-bit per sample so that the sample rate and bit rate are equal. The modulation scheme follows the waveform where the output bits indicate whether the predication value is smaller or larger than the reference sample. The encoder also maintains a reference step size, keeping the previous bits of output to determine adjustments to the step size. The step size is adjusted for every input sample processed. It is therefore very important and is adapted according to the previous inputs to a decoder. There are four key variables that determine the success of the CVSD modulation technique: the step size, the syllabic companding parameter, the decay time, and the accumulator decay factor. The feedback loop is adaptive to the extent that the loop provides a means of changing the step size depending on the previous bits. Companding is performed at a syllabic rate to extend the dynamic range of the analog input signal. Decay time is related to the length of a speech syllable and is set at 16 ms. The accumulator decay factor determines how quickly the output of the decoder returns to zero when the input is constant and is set at 0.5 ms.

In practice, the CVSD codec is used the most. This is primarily because the A-law and µ-law codecs do not tolerate data errors as well as CVSD. With no packet retransmission possible with a SCO link, and little in the way of error detection and correction (some packets have FEC), CVSD is the safest and widely used approach.

2.3 Audio measurements

The human audio frequency range is usually assumed to be from 20 Hz up to 20 kHz. For an audio signal, which is non-electrical, the first step is to convert it to an electrical signal so that it can be analyzed with instrumentation, such as an oscilloscope. A scope provides a good visual representation of the signal and shows characteristics such as loudness and pitch. The requirement for more advanced and specific audio analysis led to the introduction of dedicated audio analyzers to the market. Such equipment allows for more advanced analysis for a wide range of signals where a visual representation does not lend itself to easy or fast measurement.

In the case of *Bluetooth*, a CVSD codec operates from 200 to 3.4 kHz. The most common use of audio is with mobile phones and headsets. The speech from the user is digitally encoded by the headset or phone, and *Bluetooth* technology provides the digital wireless link between the two devices. In order to test this kind of configuration, the test equipment must be able to successfully replicate this real world scenario as best as possible. This involves factors such as initiating the *Bluetooth* link, testing other parts of the device (not just audio), the quality of the measurement and automation for manufacturing test suitability. There are a range of options as to how a device can be tested. As a result, the N4010A test set is designed to address as many of the scenarios as possible. Table 1 summarizes a range of test scenarios and will be referenced in more detail in the following section. It also helps to visualize the test setup and to avoid confusion between analog and digital references to "audio".

Depending on the device, there maybe practical limitations as to how a device can be tested. For example, a *Bluetooth* module inside a phone can only be tested if there is a suitable way of routing the *Bluetooth* signal via an antenna and the phone firmware permits this to happen. So the test setup and the device play a large part in what measurements can be made, sometimes dictated by the limitations of the device under test (DUT) being put in a suitable test mode.

The N4010A measures the following audio metrics:

1. Total harmonic distortion plus noise (THD+N)
2. SINAD
3. Frequency of the fundamental
4. Level

Distortion is something that alters a pure signal and therefore reduces the quality. Total harmonic distortion (THD) tests for the presence of a non-linearity that causes unwanted signals to be added to the input signal. A fast Fourier transform (FFT) is performed, which shows the addition of these components that are harmonically related to the input. The THD+N is similar to THD except that individual harmonics are not measured, but rather everything is added to the original input, such as harmonics and noise. This measurement must be specified alongside frequency, level, and gain for it to be meaningful.

The N4010A CVSD has a signal bandwidth of 4 kHz with a linear range between 320 Hz to 3.2 kHz and a level less than –15 dBm0. It is only suitable for low quality audio. The N4010A *Bluetooth* CVSD audio is sampled at a frequency of 8 kHz. The CVSD algorithm is a non-linear successive approximation tracking algorithm that introduces increasing distortion as the amplitude or the frequency of the signal is increased. An effect of this is that there is typically distortion above the 4 kHz signal bandwidth. When using the N4010A with audio frequencies, which are multiples of 1 kHz, the harmonic distortion components fold over and interfere when sampled at 8 kHz. There will be a "foldover" of some harmonic components above 4 kHz. For example, components at 5 kHz will appear to be the same as components at 3 kHz, but with a variable phase relationship. Therefore, this aliasing will cause variations in SINAD measurements within the range of the N4010A audio analyzer. One kilohertz, and multiples thereof, are not a good choices of frequency to make measurements. Optimum results will be achieved using frequencies that are odd multiples of 125 Hz. When using a frequency of 1.125 kHz, the signal aliases do not fall on top of the lower harmonics, so a much lower variation in SINAD occurs resulting in more stable SINAD measurements.

SINAD is a parameter that provides a quantitative measurement of the audio signal from a device. SINAD is the ratio of the total signal power level (wanted signal + noise + distortion) to the unwanted signal power (noise + distortion). The recovered audio power is the original modulating audio signal plus noise plus distortion powers from a modulated radio frequency carrier. The residual audio power is the noise-plus-distortion powers remaining after the original modulating audio signal is removed. For most practical purposes, SINAD is equal to the reciprocal of the distortion measurement. It is returned in dB by the test set.

Finally, frequency and level is also returned by the test set.

The analysis above is done by averaging results. The number of averages may range from 1 to 100. Each individual measurement is taken by doing a FFT analysis on 1,024 points (128 ms of data). The basic information extracted from each individual measurement is the total power, frequency at the peak power, peak power in the spectrum, and the power at harmonics. The measurements only start once three successive measurements have returned the same frequency measurement as the transmitted audio frequency. This supports the situation where there is a delay is in the test path such as a phone with a voice delay to allow a speak-listen mode.

If the test configuration is being changed between tests, then the new setting is only taken when the test is started. This presents no problems if frequency is changed, but if only the level is changed and the connection is maintained, then the test must be run briefly to allow any delay buffers in a DUT to fill with the revised amplitude signal. The "settling" test will still get the correct frequency and will start the measurement. The intended use case is manufacturing test where the instrument sequencer would switch modes to run tests.

3. Practical *Bluetooth* Audio Test

This section describes how the N4010A may be used with different device setups, how to configure the N4010A, and how a *Bluetooth* profile may be required to allow for audio test.

3.1 Profiles

Bluetooth profiles describe the procedures a device must adhere to in order to perform a given task (such as act as a headset, act as a wireless serial interface, act as a printer interface, stream video, etc). They ensure that different devices can work (interoperate) reliably together when performing those tasks.

For example, a wireless *Bluetooth* headset for your mobile phone may incorporate the headset profile (HSP). This profile defines how headsets and mobile phones operate together, including how to move audio signals to and from a phone, and how to control calls.

In order for two (or more) *Bluetooth* devices to connect to each other and perform a specific application, both devices must support the profile for that application. When testing an "in plastics" consumer *Bluetooth* product (e.g. a headset, mobile phone, car kit, USB network adapter, *Bluetooth* router, etc.) there is a high probability that the only way to establish a connection, and hence test the device, is through the use of a *Bluetooth* profile.

3.1.1 The headset profile

In many early *Bluetooth* products (like headsets) it was possible to get a SCO connection without using an HSP, with a product's higher level layer/firmware allowing this to happen. Now the higher level layer/firmware of many devices requires that the device they are connecting to (e.g. N4010A) must support the HSP, otherwise they will not allow a SCO connection to be made (only allowing the initial ACL connection to happen so that that profile/services information can be discovered).

N4010A Option 112 provides support for the HSP. The HSP's main objectives are:

1. To manage communication using a subset of AT commands from GSM 07.07 (such as the ability to ring, answer a call, hang up, and adjust the volume)

2. Manage the transport audio over SCO

Within the HSP a device can take one of two roles, either the audio gateway allowing for headset testing, or the headset role for phone testing.

3.1.1.1 Audio gateway role

In this role the device acts a conduit through which encoded audio will pass.

For example, a mobile phone, when used in conjunction with an audio headset, will adopt the audio gateway role. If we consider the audio routing for a received GSM call, the phone receives the GSM signal, recovers the digitally encoded audio, then passes/routes that digitally encoded signal to the *Bluetooth* interface of the phone. It is then transmitted over the air to the *Bluetooth* headset. No analog audio signals are recovered within the phone—it only passes the digitally encoded audio from the GSM decoder to the *Bluetooth* RF interface. Similarly the audio gateway passes the encoded audio received from the headset to the GSM audio encoder, which is then transmitted over the GSM RF interface.

3.1.1.1.1 Configuring the N4010A for headset testing

Under most circumstances the connection between the N4010A and the headset will be over-the-air (using antennae). For the purposes of demonstrating the use of the audio gateway role for headset test, it is assumed that the test configuration in this case is a headset using an external coupler to loopback audio to the N4010A. See Figure 1.

To ensure a reliable connection during ACL and SCO connections it is recommend to use N4010A trasmit (Tx) and receive (Rx) levels of –10 dBm. In the following steps [ ] indicate the use of hard key and the ( ) the use of a menu soft key:

1. On the N4010A front panel, press **[CONFIG]** then **(STE)**.

2. Set the **Transmit Power** and **Input Level** to **–10 dBm**. See Figure 2.

3. Make sure the headset (DUT) is powered on and is discoverable. On the N4010A front panel press **[CONFIG]**, **(EUT)**, then perform an **(Inquiry Procedure)**, **(Select & Return BDA)** on the device to which you wish to connect. See Figure 3.
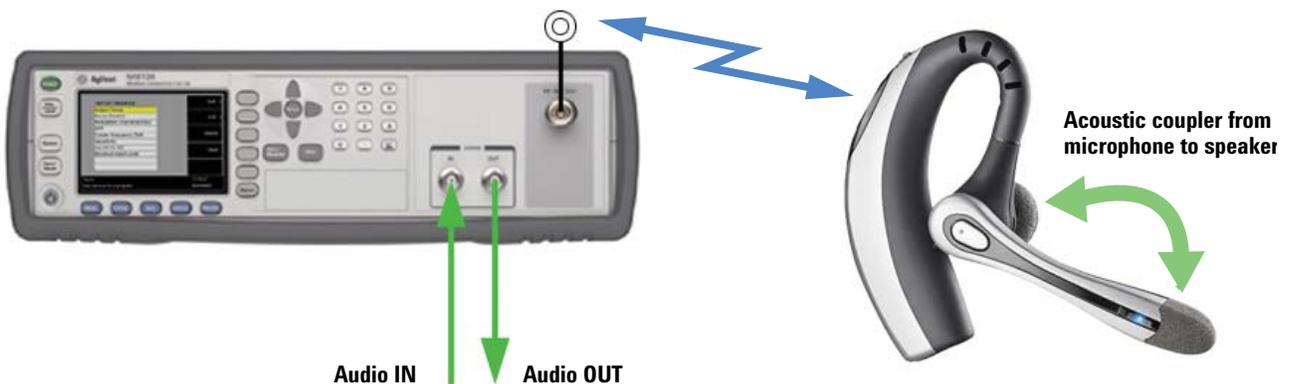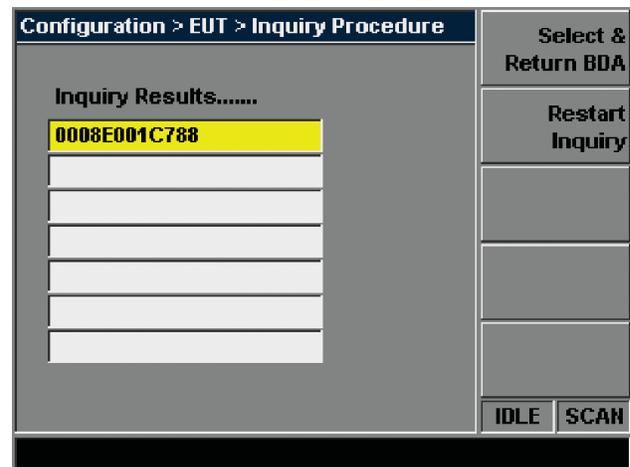
Figure 2. Configure STE

Figure 3. EUT inquiry

Figure 1. Headset test example setup

Acoustic coupler from microphone to speaker

Audio IN   Audio OUT

4. On the N4010A front panel, press **[CONFIG]**, **(Security)** and enter the correct (PIN) code for the DUT (headset)—in this example the PIN code is **'0000'**. See Figure 4.

5. On the N4010A front panel, press **[CONFIG]** and select the **Headset** profile. See Figure 5.

6. Press **(Profile)** and change the headset role to **AG** (audio gateway). See Figure 6.

7. Press **(Activate Profile)** (the status field will show **'SCAN'** meaning that is it now discoverable by other *Bluetooth* devices). See Figure 7.
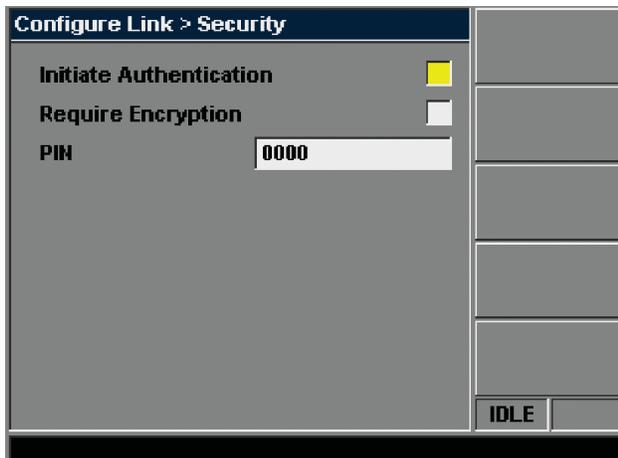
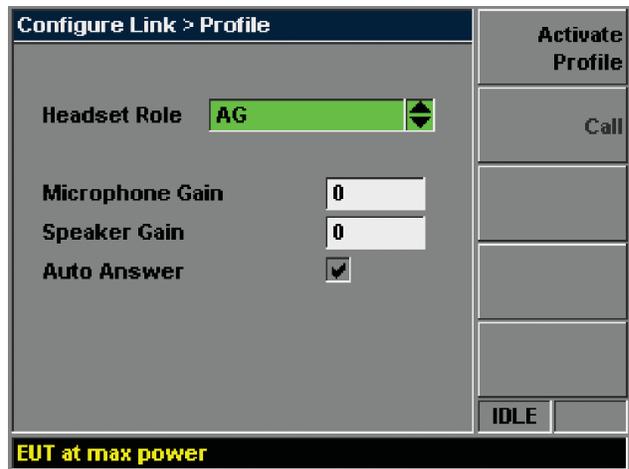**Figure 4. Security**

Figure 5. Select headset profile

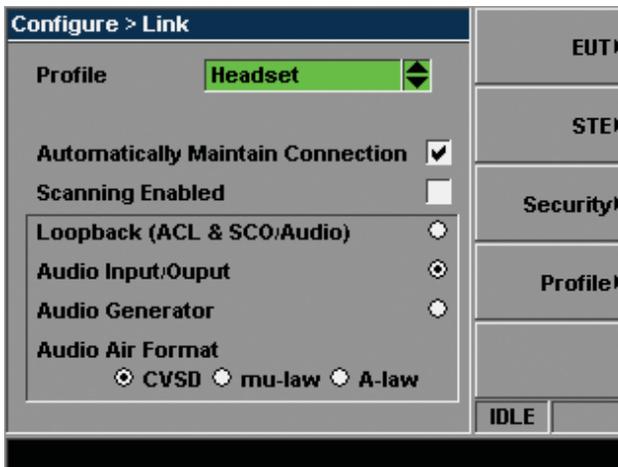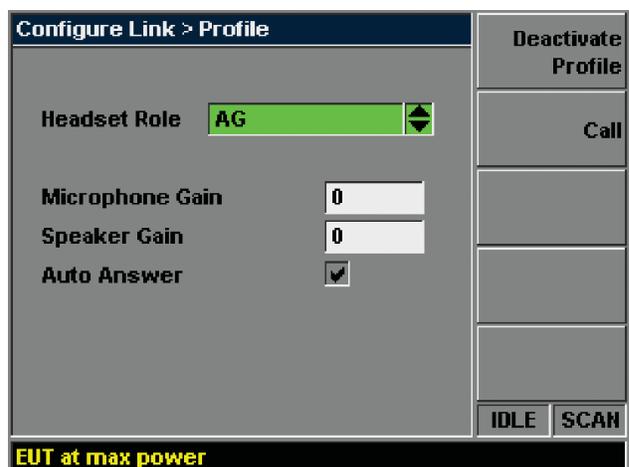**Figure 6. Select audio gateway**

Figure 7. Activate profile

8. On the N4010A press the **(Call)**. See Figure 8.

   The N4010A may briefly display **'Pairing Completed'**, then **'alerting, ring *x* of 10'** to indicate that it is calling/alerting the DUT (headset).

9. The DUT (headset) will auto-answer the call. The N4010A will briefly show **'Adding SCO Channel'** then display **'Cellular Call Active'**. The status will indicate SCOM (SCO channel with N4010A operating as master). See Figure 9.

   The connection between the N4010A and DUT (headset) has been established. It is now possible to route audio between the two devices, or perform the N4010A RF parametric SCO measurements.

10. While the call is active it is possible to change the **Microphone Gain** and **Speaker Gain**. Either press the control buttons on the DUT (except when using HCI loopback) or, enter new values in the N4010A fields. See Figure 10.

11. The call can be terminated pressing **(End Call)** on the N4010A or using the DUT (headset). The N4010A will indicate the following states:

    a. **'Cellular Call disconnected (ACLM)'** then

    b. **'RFCOMM data-link disconnected (ACLM)'** then

    c. **'RFCOMM data-link disconnected (IDLE)'**

Figure 8. Place call
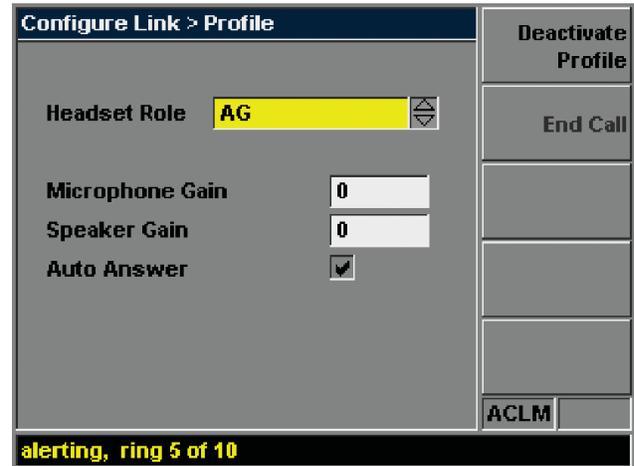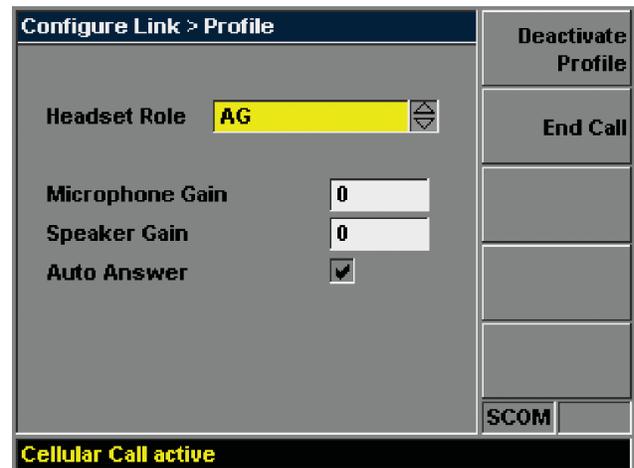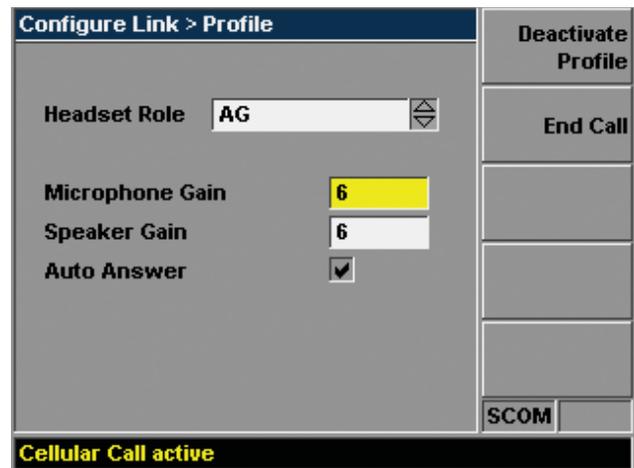

Figure 9. Cellular call active

Figure 10. Microphone and speaker settings

3.1.1.2 Headset role

In this role the device acts as the terminal that recovers or sources the analogue audio signal. The headset profile is not exclusively used for mobile phones and is commonly used to establish connections between a PC and a *Bluetooth* headset.

3.1.1.2.1 Configuring the N4010A for phone testing
In order to test a mobile phone, the DUT must use its headset profile operating in audio gateway role.

In the example below, a cellular call is established between the Agilent 8960 Wireless Communications Test Set and the mobile phone. The N4010A headset profile is activated in the headset role. The mobile phone then establishes a *Bluetooth* connection to the N4010A (in the same way it would connect to a headphone). For the purposes of this example, the N4010A audio paths are set to loopback. HCI remote loopback is used to bypass codec, hence eliminating any audio degradation caused by the N4010A. (The use of HCI remote loopback will be explained in more detail later.) An audio signal is then applied to the 8960, which transmits it to the mobile phone. The mobile phone then sends the signal to the N4010A, which then returns (loops back) it to the mobile phone and back to the 8960. The audio is recovered by the 8960. See Figure 11.

In this example no analog audio interfaces, circuits, or components are tested (i.e. the mobile phone's microphone and speaker are not tested). The routing of the audio through the phone is digitized audio (ultimately over RF). The only circuits associated with audio that are "tested" within the phone are the GSM codec, and *Bluetooth* (CSVD) codec interfaces and conversion processes.

Audio gateway testing is very similar to head set testing (see Section 3.1.1.1.1).

1. On the N4010A front panel press **[CONFIG]** then the **(STE)**.

2. Set the **Transmit Power** and **Input Level** to **–10 dBm**. See Figure 12.
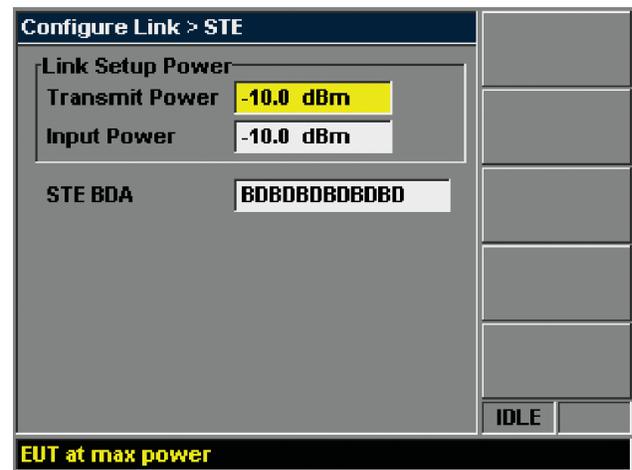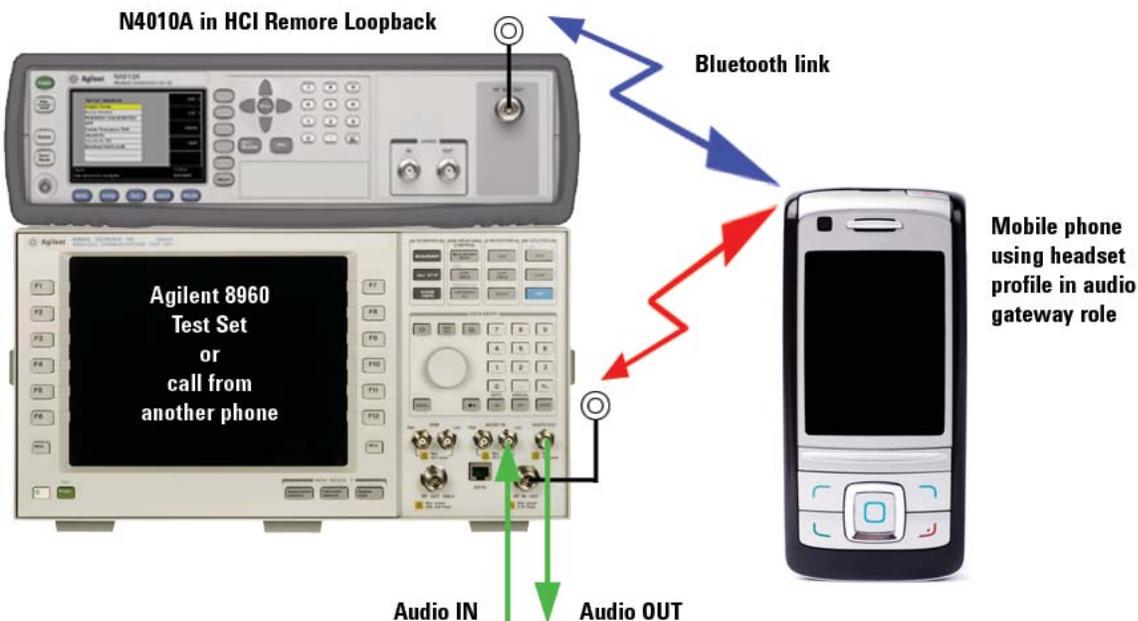
Figure 12. Setup STE

Figure 11. N4010A and 8960 test setup for audio phone testing

3. Make sure the mobile phone is powered on and is discoverable.

4. On the N4010A front panel press **[CONFIG]**. Choose the Headset profile, then press **(PROFILE)** to enter the profile menu and configure the HS role.

5. Press **(Activate Profile)**. The N4010A will show '**SCAN**'.

6. From the mobile phone perform a search for devices (i.e. an inquiry) and select the N4010A (it will be reported as '**Agilent Technologies** – **Monaco**').

7. The phone will ask you to enter the PIN code of the N4010A (the default value is '**0000**').

8. Look at the N4010A. You will see that an '**ACLS**' (ACL with N4010A acting as a slave) connection is present. It will also indicate when the authentication process happens before returning '**Pairing Complete**' if the connection is successful.

9. The N4010A is now configured/active as the headset for the DUT (mobile phone).

10. After a short while the connection status will go back to '**IDLE**' (the phone will place itself into sniff or park mode).

11. Place a call to the mobile (use either a landline or another mobile phone, or use the 8960 test set). Note that it is not possible to originate a call from the N4010A in this mode. The headset profile does not allow a headset to originate a call; only a hands-free profile would support that.

12. The N4010A will auto-answer the call.

3.2 *Bluetooth* audio test scenarios

Using the front panel of the N4010A, the configuration mode of the instrument allows for configuration of a SCO link as well as other STE and DUT parameters, such as security and profile settings. See Figure 13.

A number of test setups are possible using the N4010A test set. The N4010A Option 113 has three methods of routing audio signals between the N4010A and DUT:

1. Loopback (HCI_Remote_Loopback)

2. Audio Input/Output BNCs

3. Internal audio generator/analyzer

Out of the three audio routing methods above, there are two possible processing paths in the N4010A. The first is to use the audio codec to take data to and from the front panel BNCs. This is the basic audio I/O routing that was provided with Option 111. Option 113 now supersedes this, and also includes use of the BNC as before. The second path sources and receives data over HCI to the *Bluetooth* processor and its audio codec. N4010A audio loopback (HCI remote loopback) and the internal audio generator/ analyzer use this path. There is no "mixed" mode; either both transmit and receive signals are over HCI or both are via the front panel audio I/O BNCs. It should also be noted that the N4010A cannot support more than one channel. This means that it cannot test stereo audio unless the DUT can route individual channels.

If the test setup requires an externally-generated signal, then it must be the same frequency as setup by the N4010A transmitter due to the settling requirement. The measurements are FFT-based and work with signals that are an integer multiple of 7.8125 Hz. For example, 1375 Hz can be measured correctly. However if 1376 Hz is used, it will return 1375 Hz as the peak power frequency and with an effect on distortion.

These three routing options have the flexibility to address different customer device requirements. It is necessary to understand the types of audio loopback that may be required by the DUT (or its test fixture), and examine what type of audio functionality is available for testing using the three N4010A methods above. For example, for test of a headset, it is unlikely that a headset has the ability to internally loopback audio signals. The most probable reason for this is that such methods bypass the microphone and earpiece, which are vital components that are required for test.

For each of the following methods, the test setup is discussed in terms of the elements tested and both the STE and DUT requirements and considerations. Finally, Table 1 in Section 3.6 provides a one page summary of the options available for test as a one page guide.

Figure 13. Link configuration
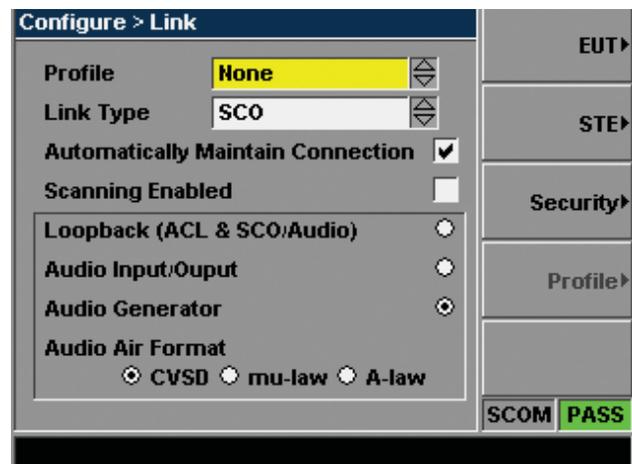
3.3 Loopback (HCI_Remote_Loopback)

Any audio signals originating from the DUT will be looped back within the N4010A (at the HCI level) and returned to the DUT. The mode is sometimes referred to as HCI_Remote_Loopback or ACL and SCO/Audio loopback. Using this method, analog audio source and analysis must be carried out through a direct connection to the DUT. This setup provides test of the audio input and output paths, and the performance of the codec to encode and decode. See Figure 14.

Note that the N4010A BNC audio input and output ports are disabled. This is indicated by the crosses through the BNC connectors, PCM codec, and CVSD transcoder in Figure 15.

The advantage of this setup is that the N4010A does not contribute any noise or distortion to audio measurements. However, a good RF Interface should be maintained between the N4010A and DUT to avoid packet errors, resulting in "pops and crackles" on the audio signal.

Caution! *This method should be used with caution. HCI_Remote_Loopback breaks the interface between the N4010A HCI layer and upper control layers. This means that any control requests issued by the N4010A will not be received by the DUT and control requests issued by the DUT will not be interpreted by the N4010A (they will just be looped back to the DUT). HCI_Remote_Loopback (ACL and SCO/Audio loopback) must be disabled in order to perform control tasks.*
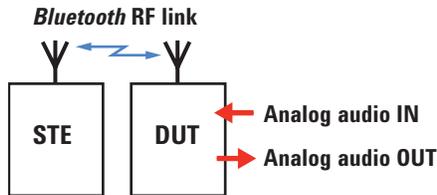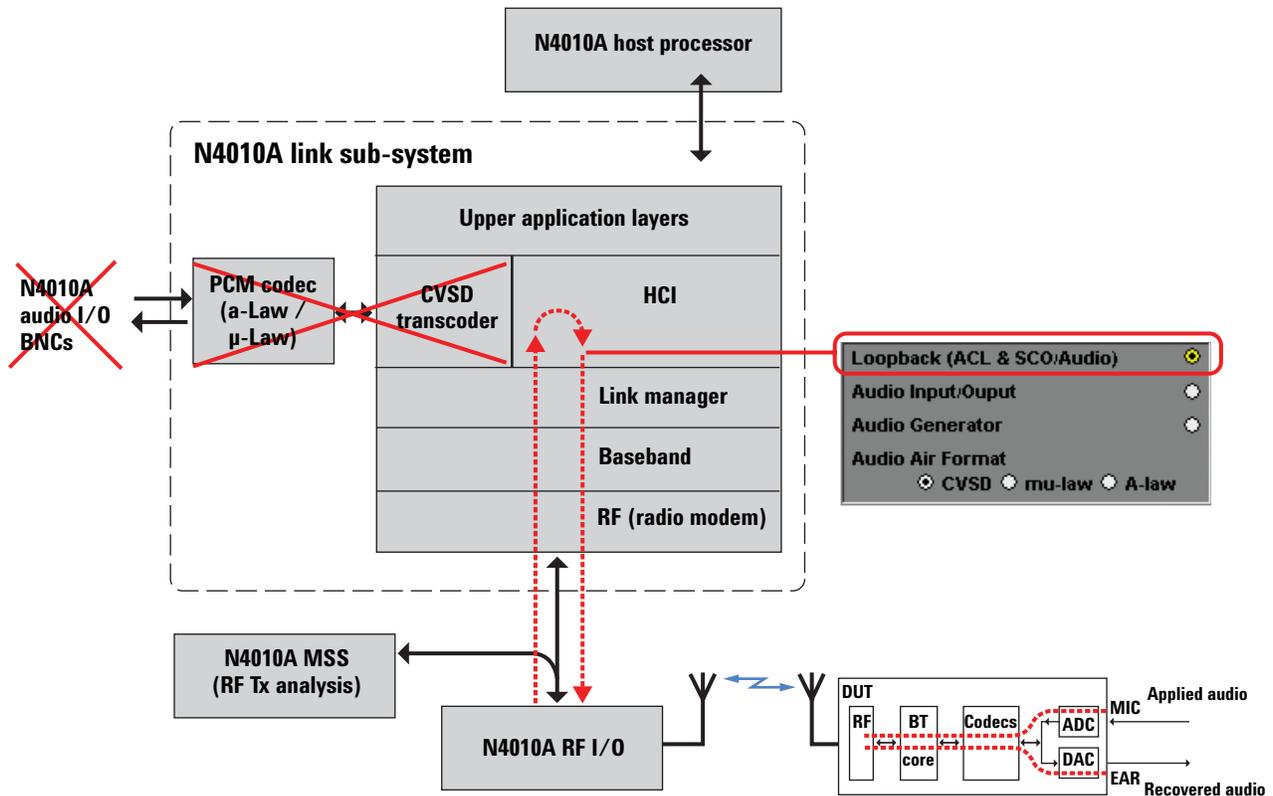
Figure 14. Loopback overview diagram

Figure 15. N4010A loopback block diagram

3.3.1 Configuring loopback audio test

1. Follow the instructions (Steps 1-8) as before to enable the headset profile if required (see Section 3.1.1.1.1). Note that in Step 5, Figure 5, HCI loopback mode must not be selected because an existing SCO connection must be active first—either select Audio Input/Output as shown in Figure 5 or Audio Generator settings shown in Figure 16.

2. Having placed a call to the DUT, it is now possible to use HCI loopback. See Figure 17.

3.3.2 Returning loopback audio test results

In HCI loopback you cannot make measurements using the N4010A test set. The N4010A is routing audio back to the device. Audio analysis is possible using an audio analyzer connected to the DUT.
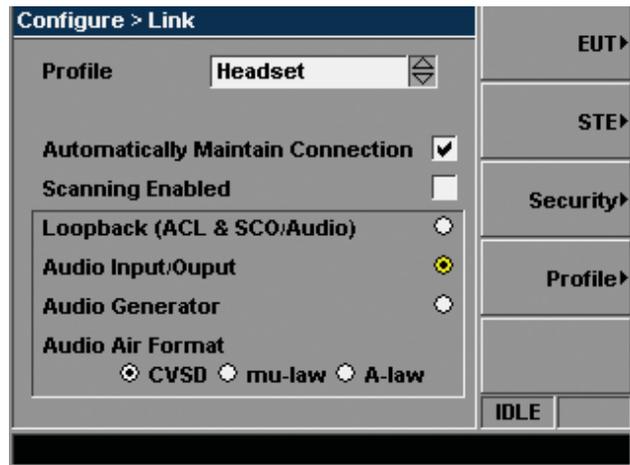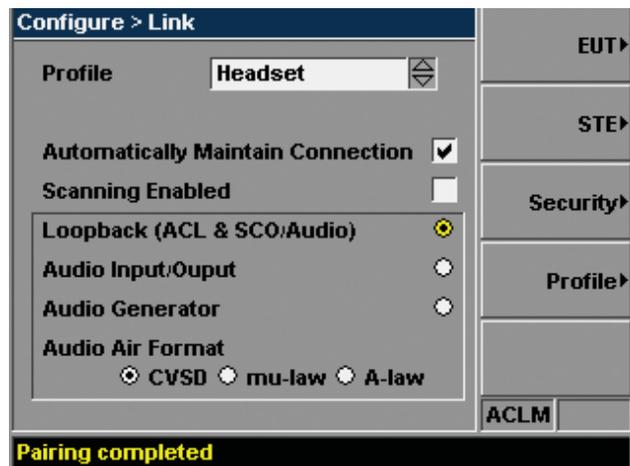
Figure 16. Configure audio routing

Figure 17. Select HCI loopback

3.3.3 Automating loopback audio test

The following programming example uses the N4010A headset profile to connect to a DUT. Audio is looped back to the DUT for an external audio analyzer to return test results.

```
/* setup link  */

LINK:TYPE SCO                                  // synchronous connection oriented method
                                                  selected for voice/audio comms
LINK:CONNect:AUTO 1                            // no automatic link disconnect
LINK:TX:POWer:LEVel -40                        // set transmit power level
LINK:RX:POWer:RANGe -25                        // set expected input power
LINK:STE:BDADdress #hBDBDBDBDBDBD              // set N4010A address
LINK:EUT:BDADdress #h000C7817FF9F              // set EUT BT device address


/* setup audio gateway */

CONF:LINK:PROFile HEADset                      // select headset profile
CONF:LINK:PROFile:HEADset:ROLE AGATeway        // select audio gateway role
CONF:LINK:PROFile:HEADset:AGATeway:SGAin 15    // set speaker gain
CONF:LINK:PROFile:HEADset:AGATeway:MGAin 15    // set microphone gain


/* setup audio parameters */

LINK:AUDio:AFORmat CVSD                        // specify CVSD encoding
LINK:AUDio:ROUTe INOUTput                      // enable audio in/out BNC connectors to setup
                                                  SCO link first
LINK:AUDio:ROUTe LOOPback                      // then loop received audio back to transmitting
                                                  EUT


/* setup DUT */

/* put DUT in pairing mode */                  // DUT commands


/* active profile & call */

LINK:PROFile:ACTivate                          //  activate headset profile & role
LINK:PROFile:HEADset:AGATeway:CALL             // open call


/* make measurements */

/* use external audio analyzer */


/* end call */

LINK:PROFile:HEADset:AGATeway:ENDCall          // close call
LINK:PROFile:DEACtivate                        // deactivate headset profile & role
```

3.4 Audio input/output

Using this method, the BNC audio input and output are enabled. Audio signals applied to the audio input are routed onto a SCO channel, applied to the RF, and transmitted to the DUT. Audio signals originating from the DUT are recovered by the N4010A and presented at the audio output BNC. Figure 18 shows how the audio input/output method works inside the N4010A in more detail. The BNC connectors on the front panel are highlighted.

The disadvantage of this approach is that the N4010A audio I/O circuits will contribute some noise and distortion to any measurement results. This can be quantified by referring to the N4010A datasheet information.

Figure 18. N4010A audio input/output block diagram
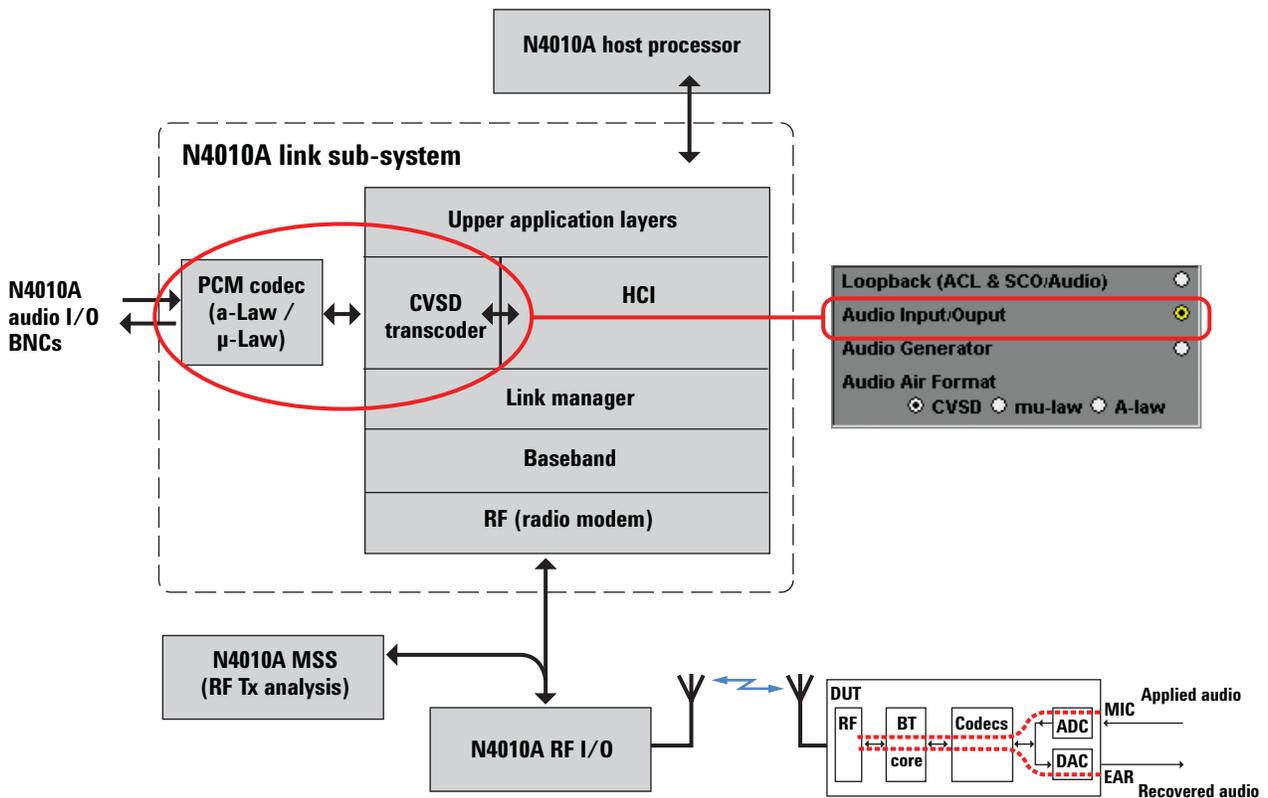
With the N4010A only routing audio signals, an external audio source and an audio analyzer is required to complete the overall test setup. Such external equipment is connected to the N4010A or the DUT depending upon the test scenario. (See Figures 19 to 22).

The setup in Figure 19 only tests the DUT audio input path and encoding of the codec. The audio source is connected to the DUT and the analyzer is connected to the audio output BNC of the N4010A.

Similarly, to test the audio output path in a device and the decoding of the codec, the audio source can be connected to the BNC of the N4010A and the audio analyzer to the audio output of the DUT. See Figure 20.

It is possible to test more effectively and make better use of the external audio source and analyzer by looping back analog audio at the DUT side. This is a common configuration in manufacturing test, shown in Figure 21.

Here the test setup uses analog audio loopback (coupling). So for a headset, the speaker, and microphone are acoustically coupled effectively using an appropriate external housing and coupler. This then means that the test setup can test the DUT analog audio input, analog audio output, codec encode, and codec decode.

The final configuration is to loopback analog audio at the N4010A side. This implies connecting the audio source and analyzer equipment to the DUT input and output. From a manu- facturing point of view this does not make much sense because the production line will require DUTs to be interchanged. The setup shown in Figure 22 is possible. However, the configuration shown in Figure 21 provides a better way. It shows external audio equipment interfacing with the STE instead.

3.4.1 Configuring audio input/output test

1. Follow the instructions as before to enable the headset profile if required (see Section 3.1.1).

2. Ensure that Audio Input/Output is selected. See Figure 23.

3.4.2 Returning audio input/output test results

In audio input/output mode you cannot make measurements using the N4010A test set; the N4010A is routing audio to/from external audio equipment. Audio analysis is possible using an audio analyzer connected to the DUT or STE (see Figures 19 to 22).

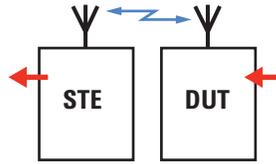**Figure 19. Audio input/output (test scenario 1) overview diagram**

Figure 20. Audio input/output (test scenario 2) overview diagram
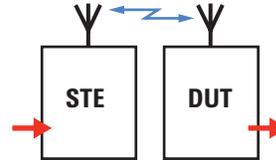

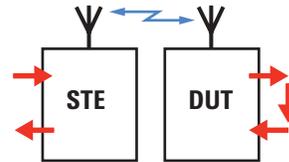**Figure 21. Audio input/output (test scenario 3) overview diagram**

Figure 22. Audio input/output (test scenario 4) overview diagram
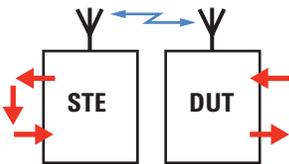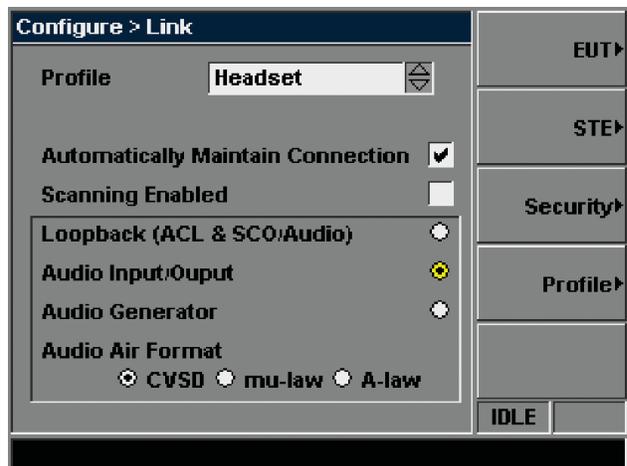

Figure 23. Select audio input/output

3.4.3 Automating input/output audio test

The following programming example uses the N4010A headset profile to connect to a DUT and route audio to external equipment using the BNC on the front panel.

```
/* setup link  */

LINK:TYPE SCO                                    // synchronous connection oriented method
                                                 selected for voice/audio comms
LINK:CONNect:AUTO 1                              // no automatic link disconnect
LINK:TX:POWer:LEVel -40                          // set transmit power level
LINK:RX:POWer:RANGe -25                          // set expected input power
LINK:STE:BDADdress #hBDBDBDBDBDBD                 // set N4010A address
LINK:EUT:BDADdress #h000C7817FF9F                 // set EUT BT device address


/* setup audio gateway */

CONF:LINK:PROFile HEADset                        // select headset profile
CONF:LINK:PROFile:HEADset:ROLE AGATeway          // select audio gateway role
CONF:LINK:PROFile:HEADset:AGATeway:SGAin 15      // set speaker gain
CONF:LINK:PROFile:HEADset:AGATeway:MGAin 15      // set microphone gain


/* setup audio parameters */

LINK:AUDio:AFORmat CVSD                          // specify CVSD encoding
LINK:AUDio:ROUTe INOUTput                        // enable audio in/out BNC connectors


/* setup DUT */

/* put DUT in pairing mode *//// DUT commands


/* active profile & call */

LINK:PROFile:ACTivate                            //  activate headset profile & role
LINK:PROFio:HEADset:AGATeway:CALL                // open call


/* make measurements */

/* use external audio analyzer */


/* end call */

LINK:PROFile:HEADset:AGATeway:ENDCall            // close call
LINK:PROFile:DEACtivate                          // deactivate headset profile & role
```

3.5 Audio generation and analysis

Using this method a tone is generated internally by the N4010A. The tone is a digital representation that is applied to the SCO channel at the HCI level. The test set's audio source is programmable in frequency and level. The signal source is a multiple of 125 Hz. This choice of steps allows easy FFT analysis and display of the exact frequency. Frequencies of 125 to 3875 Hz may be generated.

If the audio signal is looped back by the DUT (or an acoustic coupler within its test fixture) the N4010A will analyze the tone for frequency, amplitude, SINAD, and THD+N using the internal audio analyzer. Figure 24 shows the test setup in more detail. Note that

the BNC connectors are deactivated and Option 113 provides the ability to return results to the N4010A host processor.

The advantage of this method is the reduction in external test equipment requirements (i.e. analog audio source and analog audio analyzer) and connections. Software development effort can be reduced by using one instrument for audio generation and analysis, as well as being able to carry out *Bluetooth* v1.2 and v2.0 RF standard tests. Audio testing is therefore relatively simple and convenient.

When using the internal analog audio generator and analog audio analyzer, it is not possible to use the audio output BNCs on the N4010A. It is also extremely important to use frequencies that are not set at 1 kHz increments.

Figure 24. Audio generation and analysis block diagram
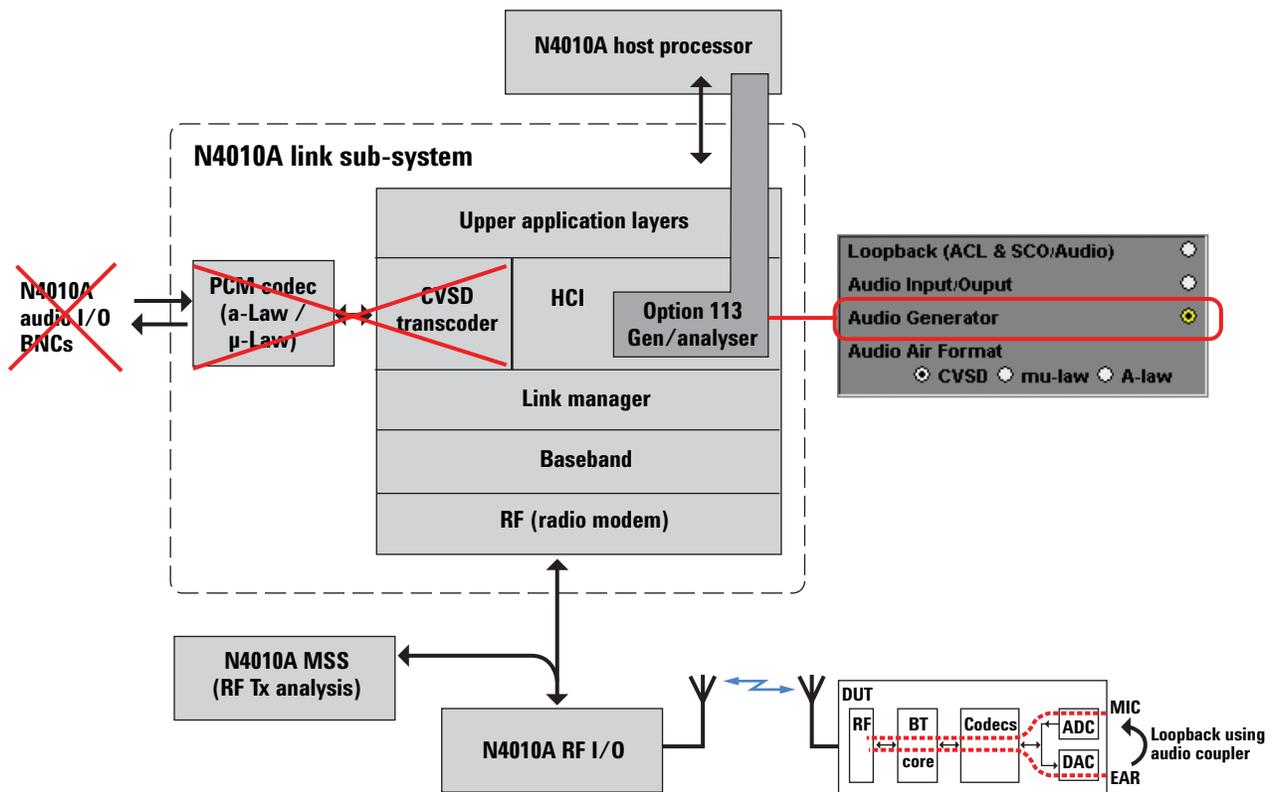
A common configuration with the audio generator and analyzer is shown in Figure 25. This setup can test the input and output analog audio paths and the encoding and decoding ability of the codec. Here the microphone and speaker of a headset can also be tested.

By looping internally in the device and not making use of analog audio paths, the test setup in Figure 26 only tests the encoding and decoding of the codec. Comparing with Figure 25, the microphone and speaker are not tested. The digital tone is generated by the STE and analyzed by the STE.

Like the previous setup, the test setup shown in Figure 27 is not testing the analog audio paths. The setup would appear to be the same as Figure 26 but it is not. Here the DUT generates the digital audio tone and the STE analyzes the output, and so only the encoding of the codec in the DUT is tested. The digital input tone from the STE to the DUT is still present but ignored by the device. However, the STE generator must be setup to match the audio generated from the DUT so that the analyzer can measure appropriately.

The setup shown in Figure 28 is the same as before except that the generated tone in now not from the device, but from an external source. Again, the digital input tone from the STE to the DUT is still present but ignored by the device. Similarly, the STE generator must be setup to match the audio generated from the DUT so that the analyzer can measure appropriately. With the analog audio being routed into the device, this test setup tests the audio input and the encoding of the codec.

Figure 25. Audio generation and analysis (test scenario 1) overview diagram
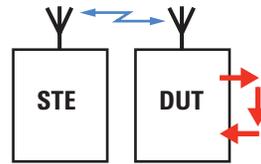

Figure 26. Audio generation and analysis (test scenario 2) overview diagram
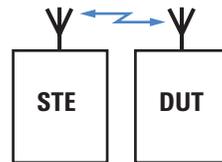

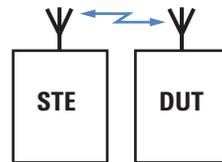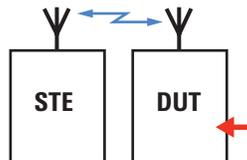**Figure 27. Audio generation and analysis (test scenario 3) overview diagram**

Figure 28. Audio generation and analysis (test scenario 4) overview diagram

3.5.1 Configuring generation and analysis audio test

1. Follow the instructions as before to enable the headset profile if required (see Section 3.1.1.1.1). However, in Step 5, choose Audio Generator and the Audio Air Format. See Figure 29.

2. Press the **[Tests]** hardkey then select '**Test Plan Active 2**'. Note that test plan 2 only does audio analysis. It is possible to choose additional test plans such as Test Plan Active 1 to add further tests, such as PER. See Figure 30.

3. Press the (**Edit**) softkey. See Figure 31.

4. Change tone frequency and tone level to those you wish to test. See Figure 32.

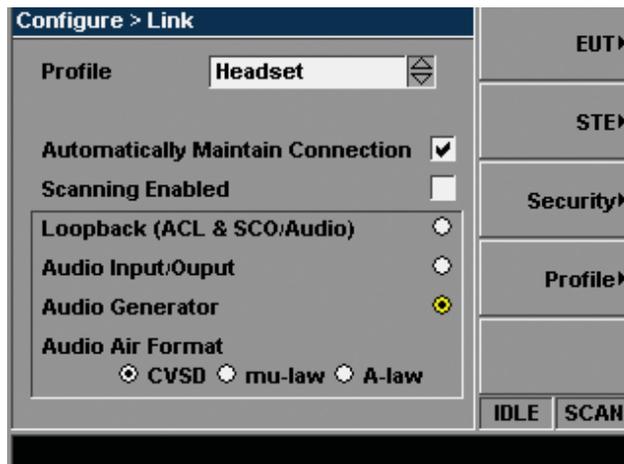**Figure 29. Select audio generator/analyzer mode**

Figure 31. Edit Audio test parameters
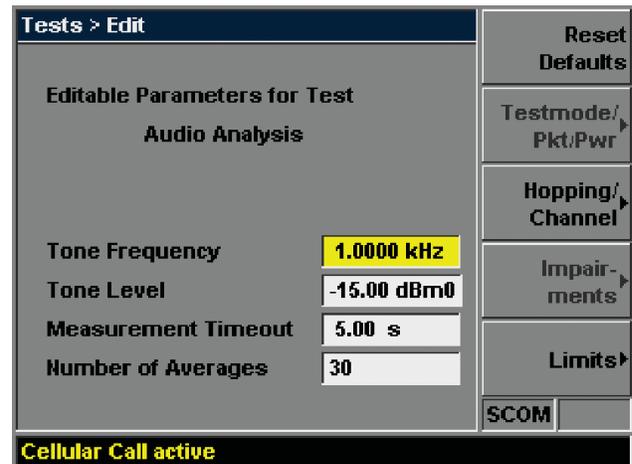
Figure 30. Select test plan

A SCO connection must be present before audio tests can be performed

Figure 32. Change tone level and tone frequency

5. If using the headset profile it may also be necessary to remotely adjust the DUT's speaker and microphone gain. See Figure 33.

3.5.2 Returning loopback audio test results

6. Press the [**RUN**] key. The test will take a few seconds to execute, then will show a PASS (or FAIL) result. See Figure 34.

7. Press the (**Detailed Results**) softkey, and select the (**Freq Analysis**) menu. See Figure 35.

8. The select the (**Dist Analysis**) menu. See Figure 36.

Figure 33. Configure speaker and microphone gain

Figure 35. Frequency analysis

Figure 34. Run test

Figure 36. Distortion analysis

3.5.3 Automating generation and analysis audio test

The following programming example uses the N4010A headset profile to connect to an DUT, generate an audio tone, analyze audio, and return results.

```
/* setup link  */

LINK:TYPE SCO                                    // synchronous connection oriented method
                                                 //    selected for voice/audio comms
LINK:CONNect:AUTO 1                              // no automatic link disconnect
LINK:TX:POWer:LEVel -40                          // set transmit power level
LINK:RX:POWer:RANGe -25                          // set expected input power
LINK:STE:BDADdress #hBDBDBDBDBDBD                // set N4010A address
LINK:EUT:BDADdress #h000C7817FF9F                // set EUT BT device address


/* setup audio gateway */

CONF:LINK:PROFile HEADset                        // select headset profile
CONF:LINK:PROFile:HEADset:ROLE AGATeway          // select audio gateway role
CONF:LINK:PROFile:HEADset:AGATeway:SGAin 15      // set speaker gain
CONF:LINK:PROFile:HEADset:AGATeway:MGAin 15      // set microphone gain


/* setup audio parameters */

LINK:AUDio:AFORmat CVSD                          // specify CVSD encoding
LINK:AUDio:ROUTe GENerator                       // enable internal audio generator & analyzer
LINK:CONFigure:AUDio:FREQuency 1125              // set audio frequency
LINK:CONFigure:AUDio:TIMeout 5                   // set length of time to detect audio tone
LINK:CONFigure:AUDio:LEVel -15                   // set output level of audio tone for generator
LINK:CONFigure:AUDio:AVERage 30                  // number of measurement results to be averaged


/* select tests */

SEQ:NORMal:ACTive 2                              // select test plan


/* setup DUT */

/* put DUT in pairing mode */                    // DUT commands


/* active profile & call */

LINK:PROFile:ACTivate                            // activate headset profile & role
LINK:PROFile:HEADset:AGATeway:CALL               // open call


/* initiate measurements */

INITiate


/* fetch results */

… start loop …
SEQ:DONE? AUD                                    // sequence of tests complete?
… if returns 1 then fetch results and exit loop …
FETCh:AUDio:FREQuency?
… read result over I/O …
FETCh:AUDio:LEVel?
… read result over I/O …
FETCh:AUDio:SINad?
… read result over I/O …
FETCh:AUDio:DISTortion?
… read result over I/O …


/* end call */

LINK:PROFile:HEADset:AGATeway:ENDCall            // close call
LINK:PROFile:DEACtivate                          // deactivate headset profile & role
```

3.6 *Bluetooth* audio test scenarios—One page summary

Table 1 summarizes how a device may be tested using the N4010A in loopback, audio input/output, and audio generation and analysis modes.

Table 1. *Bluetooth* Audio Option 113 test scenarios (CVSD)

N4010A measurement configuration	Loopback (ACL and SCO/Audio)	Audio input/output		Audio generation and analysis			
Overview	Audio applied and recovered from the DUT. N4010A provides loopback	Multiple configurations supported. For separate input and output analysis, audio can be applied to DUT and recovered by audio analyzer at the N4010A. Audio applied to N4010A and recovered by audio analyzer at the DUT. Audio can also be applied and recovered at the N4010A with audio loopback at the DUT using audio coupling		Audio applied and generated by the N4010A with loopback by the DUT		Audio generated by the DUT and analyzed by STE	Audio routed by the DUT and analyzed by STE
Audio source	External audio generator	External audio generator		N4010A internal audio generator[4]		DUT	External audio generator
Audio analysis	External audio analyzer	External audio analyzer		N4010A internal audio analyzer[4]			
Elements tested	1. Audio input 2. Audio output 3. Codec encode 4. Codec decode	1. Audio input 2. Codec encode	1. Audio output 2. Codec decode	1. Audio input 2. Audio output 3. Codec encode 4. Codec decode	1 Codec encode 2. Codec decode	1 Codec encode	1. Audio input 2. Codec encode
DUT requirements	N/A	N/A	N/A	Analog audio loopback (coupling)	Internal codec loopback	Internal audio generation (output) enabled. Audio input from STE disabled	External audio enabled. Audio input from STE disabled
Basic setup[1]	[block diagram: STE, DUT]	[block diagram: STE, DUT]	[block diagram: STE, DUT]	[block diagram: STE, DUT]	[block diagram: STE, DUT]	[block diagram: STE, DUT]	[block diagram: STE, DUT]
DUT types	Headset[4] and phone[2]	Headset[4] and phone[2]		Headset[4]		Phone[2]	Headset[4] and phone[2]
Test set considerations	HCI control commands	• Noise and distortion from I/O • BNC connections required to connect external equipment.		• Loopback I/O (using BNC) • Noise and distortion from I/O		• Frequency must not be set to multiple 1 kHz increments. • N4010A audio I/O (BNC) cannot be used	• STE not analyzing audio looped back from DUT • STE generator must be setup to match audio generated/routed from DUT/external source so that analyzer is setup correctly.

1. Basic block diagram showing analog audio input and output flow to/from the N4010A and device. SCO link setup between STE and DUT.

2. Provided microphone and speaker functions can be connected and routed to/from a *Bluetooth* link on a phone. May require N4010A Option 112 headset profile in audio headset (HS) mode.

3. No external measurement equipment is required because the internal audio generator and analyzer is being used inside the N4010A.

4. May require N4010A Option 112 headset profile in audio gateway (AG) mode.

4. Conclusion

With the growth in *Bluetooth* technology, audio test has become a common manufacturing test incorporated into *Bluetooth* test plans. Alongside RF parametric tests, audio testing requirements must be understood and integrated into test setups.

The N4010A provides three different ways to route audio to and from DUT. This provides flexibility in addressing a range of device types. The N4010A's internal audio generator and analyzer eliminate the need for external equipment and reduce cost. Measurements of THD+N, SINAD, frequency, and level provide a functional check of device performance. An appropriate setup can test input and output analog audio paths and the ability of the codec to encode and decode. In addition, audio tests can also check the functionality of microphones and speakers.

5. Appendix: 0 dBm0

In accordance to ITU specification G.711, a 775 mVrms (0 dBm) analog sine wave input voltage is translated to 0 dBm0 digital CVSD transmit signal and 0 dBm0 sine wave CVSD receive signal is output as 775 mVrms (0 dBm) analog voltage.

6. Glossary

ACL	Asynchronous connectionless link
ADPCM	Adaptive differential pulse code modulation
BNC	Bayonet Neill-Concelman connector
CVSD	Continuous variable slope delta
DM	Delta modulation
DPCM	Differential pulse code modulation
DUT	Device under test
FEC	Forward error correction
FFT	Fast Fourier transform
GSM	Global system for mobile communications
HCI	Host controller interface
HSP	Headset profile
I/O	Input/Output
ITU-T	International Telecommunication Union
MSS	Measurement sub-system
PER	Packet error rate
PC	Personal computer
PCM	Pulse code modulation
RF	Radio rrequency
Rx	Receiver/Receive
SCO	Synchronous connection oriented
SINAD	Signal including noise and distortion
STE	Standard test equipment
THD	Total harmonic distortion
THD+N	Total harmonic distortion plus noise
Tx	Tranmitter/Transmit
USB	Universal serial bus

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