Hands-Free Profile

***Bluetooth*® Profile Specification**

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Abstract:

The Hands-Free Profile (HFP) specification defines a set of functions such that a Mobile Phone can be used in conjunction with a Hands-Free device (e.g., installed in the car or represented by a wearable device such as a headset), with a Bluetooth Link providing a wireless means for both remote control of the Mobile Phone by the Hands-Free device and voice connections between the Mobile Phone and the Hands-Free device.

Version History

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Change History

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v1.0 to v1.5	New features: • Respond and Hold • Subscriber Number Information • Enhanced Call Status • Enhanced Call Controls Incorporated errata 13, 261, 317, 549, 550, 575, 586, 635, 706, 731, 746, 819, 820, 821, 822, and 823.
v1.5 to v1.6	New features: • Wide Band Speech Codec • Codec Negotiation • Individual Indicator Activation Incorporated errata 913, 1859, 1868, 1872, 1878, 1934, 1958, 1989, 2037, 2043, 2144, 2146, 2209, 2211, 2259, 2276, 2286, 2459, 2484, 2713, 2716, 2742, 2855, 3090, 3152, 3688, 3816, and 3910.
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v1.7.2 to v1.8	New feature: Enhanced Voice Recognition Activation Incorporated erratum E11729.
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1 Introduction

1.1 Scope

This document defines the protocols and procedures that shall be used by devices implementing the Hands-Free Profile. The most common examples of such devices are in-car Hands-Free units used together with cellular phones, or wearable wireless headsets.

The profile defines how two devices supporting the Hands-Free Profile shall interact with each other on a point-to-point basis.

An implementation of the Hands-Free Profile typically enables a headset, or an embedded hands-free unit to connect, wirelessly, to a cellular phone for the purposes of acting as the cellular phone's audio input and output mechanism and allowing typical telephony functions to be performed without access to the actual phone.

1.2 Profile Dependencies

In Figure 1.1, the Bluetooth profile structure and the dependencies of the profiles are depicted. A profile is dependent upon another profile if it re-uses parts of that profile, by explicitly referencing it. Dependency is illustrated in Figure 1.1.

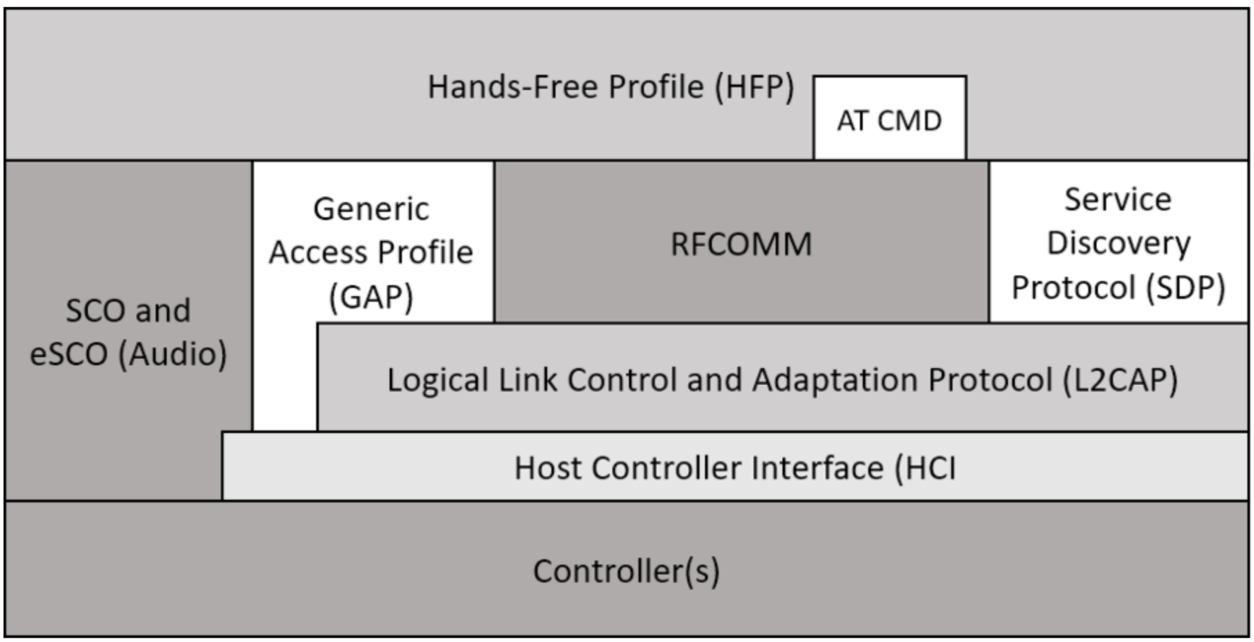

Figure 1.1: Bluetooth Profiles

As indicated in Figure 1.1, the Hands-Free Profile is dependent upon both RFCOMM [4] and the Generic Access Profile (Volume 3, Part C in [1]). Details are provided in Sections 6 (RFCOMM) and 7 (Generic Access Profile).

1.3 Symbols and Conventions

1.3.1 Requirement Status Symbols

In this document, the following symbols are used:

- "M" for mandatory to support
- "O" for optional to support
- "X" for excluded (used for capabilities that may be supported by the device, but the Hands-Free Profile shall not use these capabilities)
- "C" for conditional to support
- "N/A" for not applicable (in the given context this capability is not defined)

Some capabilities or features (identified as “X”), mandated according to the relevant Bluetooth specifications, are excluded with respect to this profile because they might degrade the operation of devices in the particular use case. Therefore, features or capabilities labeled “X” shall never be activated while operating in a use case where they are labeled as such.

1.3.2 Naming Conventions

In this document, the following naming conventions are used:

Where “Core Specification” is said it refers to the Bluetooth Core Specification Version 4.2 or later adopted by the Bluetooth SIG.

Where “LMP link” is said, it means a Link Manager (LM) level link over which only Link Manager Protocol (LMP) commands are conveyed.

Where “RFCOMM connection” is said, it means the presence of an emulated serial port as specified in [4].

- Where “Service Level Connection” is said, it means a synchronized high-level protocol connection involving a portion of the protocol stack. In this specific case, it refers to the presence of a RFCOMM connection, and assumes that the HF (Hands-Free unit) has synchronized itself to the state of the AG (Audio Gateway) using the specified Service Level Connection initialization procedure.
- Where “Service Level Connection initialization” is said, it means the execution of the set of AT commands and responses specified by the profile necessary to synchronize the state of the HF with that of the AG.
- Where “Service Level Connection establishment” is said, it means the combined process of establishing the RFCOMM connection, as well as the necessary device synchronization using Service Level Connection initialization.
- Where “Synchronous Connection” is said, it means a SCO or eSCO logical link intended for supporting a full duplex Audio Connection.
- Where “Audio Connection” is said, it means a Synchronous Connection including the means to provide a complete audio path between two devices assuming roles within this profile.
- Where “Codec Connection” is said, it means a Synchronous Connection established after profile level negotiation of Codec and related configuration.

- Where “Wide Band Speech Connection” is said, it means a Codec Connection where the media being transported consists of encoded frames derived from speech (or other audio) sampled at 16 kHz. The format of the media transported over the Synchronous Connection shall be negotiated during the establishment of the Codec Connection.
- Where “mSBC” or “Modified SBC” is said, it means a modified version of the A2DP SBC codec that is used as the mandatory codec for Wide Band Speech Connections. Section 5.7.4 contains a complete definition of the modifications which comprise mSBC.
- Where “Super Wide Band Speech Connection” is stated, it means a Codec Connection where the media being transported consists of encoded frames derived from speech (or other audio) sampled at 32 kHz. The format of the media transported over the Synchronous Connection shall be negotiated during the establishment of the Codec Connection.
- Where “LC3-SWB” is stated, it means the LC3 codec that is used as the mandatory codec for Super Wide Band Speech Connections. The complete definition of the codec can be found in the LC3 Specification [10].
- Where “incoming call” is said, it means a call connection in the direction “Phone Network=>AG”, such that it is initiated by the Network to which the AG is attached.
- Where ‘outgoing call’ is said, it means a call connection in the direction ”AG=>Phone Network”, such that it is initiated by the AG towards the Network to which it is attached.
- Where ‘legacy device’ is said, it means a device that is compatible with an earlier version, v0.96, v1.0 or v1.5, of this specification; see Section 5.3.2.

1.3.3 Signaling Diagram Conventions

Figure 1.2 shows conventions used to describe procedures in signaling diagrams.

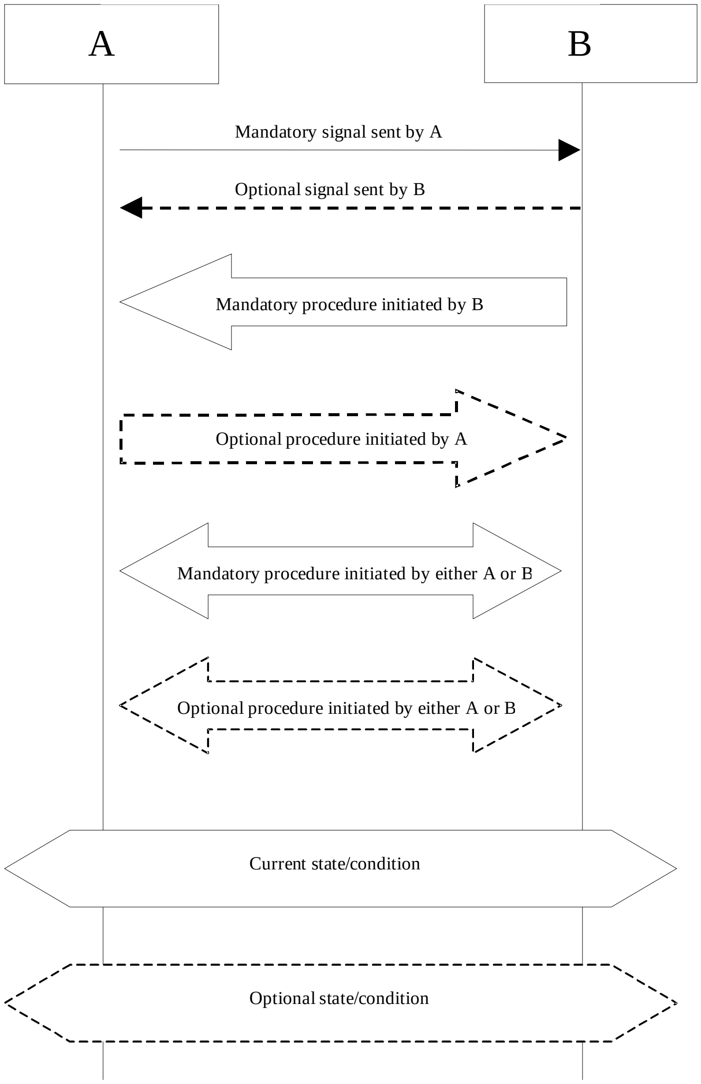

Figure 1.2: Conventions used in signaling diagrams

1.4 Language

1.4.1 Language conventions

In the development of a specification, the Bluetooth SIG has established the following conventions for use of the terms *"shall"*, *"shall not"*, "*should*", *"should not"*, "*may*", *"must"*, and *"can"*. In this Bluetooth specification, the terms in Table 1.1 have the specific meanings given in that table, irrespective of other meanings that exist.

Term	Definition
shall	—used to express what is required by the specification and is to be implemented exactly as written without deviation
shall not	—used to express what is forbidden by the specification
should	—used to express what is recommended by the specification without forbidding anything
should not	—used to indicate that something is discouraged but not forbidden by the specification
may	—used to indicate something that is permissible within the limits of the specification
must	—used to indicate either: 1. an indisputable statement of fact that is always true regardless of the circumstances 2. an implication or natural consequence if a separately-stated requirement is followed
can	—used to express a statement of possibility or capability

Table 1.1: Language conventions terms and definitions

1.4.1.1 Implementation alternatives

When specification content indicates that there are multiple alternatives to satisfy specification requirements, if one alternative is explained or illustrated in an example it is not intended to limit other alternatives that the specification requirements permit.

1.4.1.2 Discrepancies

It is the goal of Bluetooth SIG that specifications are clear, unambiguous, and do not contain discrepancies. However, members can report any perceived discrepancy by filing an erratum and can request a test case waiver as appropriate.

Certain terms used in this specification have been updated and are no longer used by Bluetooth SIG. For a list of terms that have been updated and their replacement terms, see the Appropriate Language Mapping Tables [11].

1.4.2 Reserved for Future Use

Where a field in a packet, Protocol Data Unit (PDU), or other data structure is described as "Reserved for Future Use" (irrespective of whether in uppercase or lowercase), the device creating the structure shall set its value to zero unless otherwise specified. Any device receiving or interpreting the structure shall ignore that field; in particular, it shall not reject the structure because of the value of the field.

Where a field, parameter, or other variable object can take a range of values, and some values are described as "Reserved for Future Use," a device sending the object shall not set the object to those values. A device receiving an object with such a value should reject it, and any data structure containing it, as being erroneous; however, this does not apply in a context where the object is described as being ignored or it is specified to ignore unrecognized values.

When a field value is a bit field, unassigned bits can be marked as Reserved for Future Use and shall be set to 0. Implementations that receive a message that contains a Reserved for Future Use bit that is set to 1 shall process the message as if that bit was set to 0, except where specified otherwise.

The acronym RFU is equivalent to Reserved for Future Use.

1.4.3 Prohibited

When a field value is an enumeration, unassigned values can be marked as “Prohibited.” These values shall never be used by an implementation, and any message received that includes a Prohibited value shall be ignored and shall not be processed and shall not be responded to.

Where a field, parameter, or other variable object can take a range of values, and some values are described as “Prohibited,” devices shall not set the object to any of those Prohibited values. A device receiving an object with such a value should reject it, and any data structure containing it, as being erroneous.

“Prohibited” is never abbreviated.

2 Profile Overview

2.1 Protocol Stack

Figure 2.1 shows the protocols and entities used in this profile.

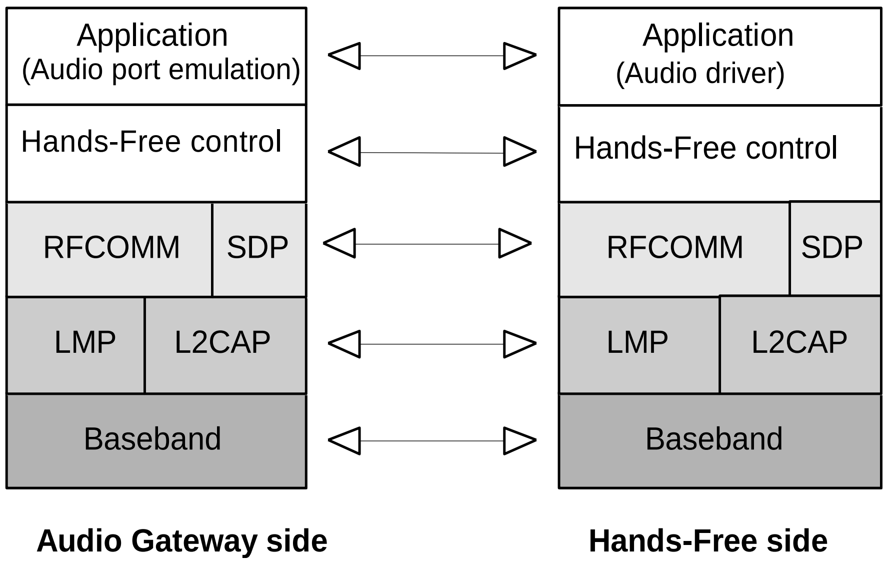

Figure 2.1: Protocol stack

The Baseband, LMP and L2CAP are the OSI layer 1 and 2 Bluetooth protocols. RFCOMM is the Bluetooth serial port emulation entity. SDP is the Bluetooth Service Discovery Protocol. See [1] for more details on these topics.

Hands-Free control is the entity responsible for Hands-Free unit specific control signaling; this signaling is AT command based.

Although not shown in the model above, it is assumed by this profile that Hands-Free Control has access to some lower layer procedures (for example, Synchronous Connection establishment).

The audio port emulation layer shown in Figure 2.1 is the entity emulating the audio port on the Audio Gateway, and the audio driver is the driver software in the Hands-Free unit.

2.2 Configuration and Roles

Figure 2.2 shows typical configurations of devices for which the Hands-Free Profile is applicable:

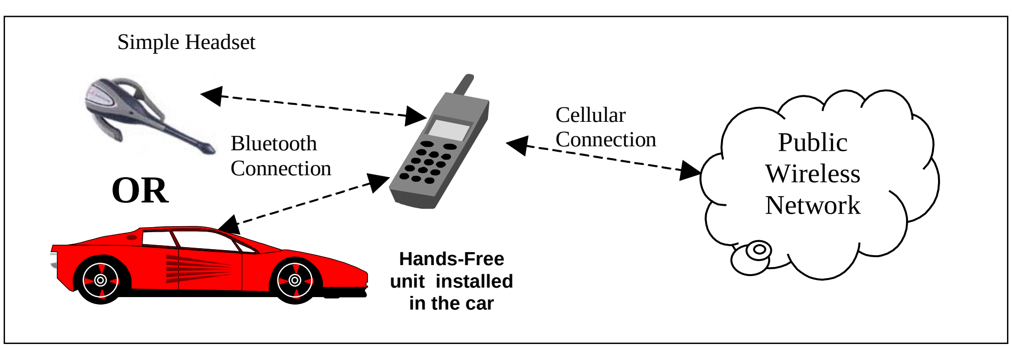

Figure 2.2: Typical Hands-Free Use

The following roles are defined for this profile:

Audio Gateway (AG) – This is the device that is the gateway of the audio, both for input and output. Typical devices acting as Audio Gateways are cellular phones.

Hands-Free unit (HF) – This is the device acting as the Audio Gateway's remote audio input and output mechanism. It also provides some remote control means.

These terms are used in the rest of this document to designate these roles.

2.3 User Requirements and Scenarios

The following rules apply to this profile:

a) The profile states the mandatory and optional features when the "Hands-Free Profile" is active in the Audio Gateway and the Hands-Free unit.

b) The profile mandates the usage of CVSD (see Volume 2, Part B, Section 9 in [1]) for transmission of audio (over the Bluetooth link). The resulting audio is monophonic, with a quality that, under normal circumstances, does not have perceived audio degradation.

c) Between the Hands-Free unit and the Audio Gateway, only one Audio Connection per Service Level Connection at a time is supported.

d) Both the Audio Gateway and the Hands-Free unit may initiate Audio Connection establishment and release. Valid speech data shall exist on the Synchronous Connection in both directions after the Audio Connection is established. Since it is only the AG that knows if Wide Band Speech or Super Wide Band Speech should be used, it should always be the AG that establishes the Synchronous Connection with the required codec.

e) Whenever an "Audio Connection" exists, a related "Service Level Connection" shall also exist.

f) The presence of a "Service Level Connection" shall not imply that an "Audio Connection" exists. Releasing a "Service Level Connection" shall also release any existing "Audio Connection" related to it.

2.4 Profile Fundamentals

If an LMP link is not already established between the Hands-Free unit and the Audio Gateway, the LMP link shall be set up before any other procedure is performed.

There are no fixed Central or Peripheral roles in the profile.

The Audio Gateway and Hands-Free unit provide serial port emulation. For the serial port emulation, RFCOMM (see [4]) is used. The serial port emulation is used to transport the user data including modem control signals and AT command from the Hands-Free unit to the Audio Gateway. The AT commands are parsed by the Audio Gateway and responses are sent to the Hands-Free unit via the Bluetooth serial port connection.

2.5 Conformance

Each capability of this specification shall be supported in the specified manner. This specification may provide options for design flexibility, because, for example, some products do not implement every portion of the specification. For each implementation option that is supported, it shall be supported as specified.

3 Application Layer

This section describes the feature requirements for units complying with HFP.

Table 3.1 shows the feature requirements for this profile.

	Feature	**Support in HF**	**Support in AG**
1.	Connection management	M	M
2.	Phone status information	M(note 1)	M
3.	Audio Connection handling	M	M
4	Accept an incoming voice call	M	M
5.	Reject an incoming voice call	M	O
6.	Terminate a call	M	M
7.	Audio Connection transfer during an ongoing call	M	M
8.	Place a call with the phone number supplied by the HF	O	M
9.	Place a call using memory dialing	O	M
10.	Place a call to the last number dialed	O	M
11.	Call waiting notification	O	M
12.	Three-way calling	O(note 2)	O(note 3)
13.	Calling Line Identification (CLI)	O	M
14.	Echo canceling (EC) and noise reduction (NR)	O	O
15.	Voice recognition activation	O	O
16.	Attach a Phone number for a voice tag	O	O
17.	Ability to transmit DTMF codes	O	M
18.	Remote audio volume control	O	O
19.	Respond and Hold	O	O
20.	Subscriber Number Information	O	M
21a.	Enhanced Call Status	O	M
21b.	Enhanced Call Controls	O	O
22.	Individual Indicator Activation	O	M

	Feature	**Support in HF**	**Support in AG**
23	Wide Band Speech	O	O
24	Codec Negotiation	O(note 4)	O(note 4)
25	HF Indicators	O	O
26	Super Wide Band Speech	O	O
Note 1: The HF shall support at least the two indicators "service" and "call". Note 2: If "Three-way calling" is supported by the HF, it shall support AT+CHLD values 1 and 2. The HF may additionally support AT+CHLD values 0, 3 and 4. Note 3: If "Three-way calling" is supported by the AG, it shall support AT+CHLD values 1 and 2. The AG may additionally support AT+CHLD values 0, 3, and 4. Note 4: If Wide Band Speech or Super Wide Band Speech is supported, Codec Negotiation shall also be supported.			

Table 3.1: Application layer procedures

Table 3.2 maps each feature to the procedures used for that feature. All procedures are mandatory if the feature is supported.

	Feature	**Procedure**	**Ref.**
1.	Connection management	Service Level Connection establishment	4.2
		Service Level Connection release	4.3
2.	Phone status information	Transfer of Registration Status Transfer of Signal Strength Indication Transfer of Roaming Status Indication Transfer of Battery Level Indication Query of Operator Selection Extended Audio Gateway Error Codes Transfer of Call, Call Setup and Call Held Status	4.4 4.5 4.6 4.7 4.8 4.9 4.10
3.	Audio Connection handling	Audio Connection set up Audio Connection release Codec Connection setup	4.11 4.12
4.	Accept an incoming voice call	Answer an incoming call	4.13
5.	Reject an incoming voice call	Reject an incoming call	4.14
6.	Terminate a call	Terminate a call process	4.15
7.	Audio Connection transfer during an ongoing call	Audio Connection transfer towards the HF Audio Connection transfer towards the AG	4.16 4.17
8.	Place a call with the phone number supplied by the HF	Place a call with the phone number supplied by the HF	4.18
9.	Place a call using memory dialing	Memory dialing from the HF	4.19
10.	Place a call to the last number dialed	Last number re-dial from the HF	4.20
11.	Call waiting notification	Call waiting notification activation	4.21
12.	Three-way calling	Three-way call handling	4.22
13.	Calling Line Identification (CLI)	Calling Line Identification (CLI) notification	4.23

	Feature	Procedure	Ref.
14.	Echo canceling (EC) and noise reduction (NR)	HF requests turning off the AG's EC and NR	4.24
15.	Voice recognition activation	Voice recognition activation	4.25
16.	Attach a phone number for a voice tag	Attach a voice tag to a phone number	4.27
17.	Ability to transmit DTMF codes	Transmit DTMF code	4.28
18.	Remote audio volume control	Remote audio volume control Volume level synchronization	4.29
19.	Response and Hold	Query response and hold status Put an incoming call on hold from HF Put an incoming call on hold from AG Accept a held incoming call from HF Accept a held incoming call from AG Reject a held incoming call from HF Reject a held incoming call from AG Held incoming call terminated by caller	4.30 4.30 4.30 4.30 4.30 4.30 4.30 4.30
20.	Subscriber Number Information	Subscriber Number Information	4.31
21a.	Enhanced Call Status	Query Call List	4.32
21b.	Enhanced Call Control	Release Specified Call Private Consult Mode	4.33 4.33
22.	Individual Indicator Activation	Indicators Activation and Deactivation	4.34
23	Wide Band Speech	Wide Band Speech	10 11 12
24	Codec Negotiation	Codec Negotiation	4.11
25	HF Indicators	HF Indicators	4.35
26	Super Wide Band Speech	Super Wide Band Speech	Appendix E

Table 3.2: Application layer feature to procedure mapping

Table 3.3 shows the codec requirements for this profile.

	Codec	Support in HF	Support in AG
1.	CVSD	M	M
2.	mSBC	C.1	C.1
3.	LC3-SWB	C.2	C.2
C.1: Mandatory if Wide Band Speech is supported, or excluded otherwise C.2: Mandatory if Super Wide Band Speech is supported, or excluded otherwise			

Table 3.3: Requirements on supported codecs

Table 3.4 shows a summary of the mapping of codec requirements on link features for this profile.

	Feature	Support in HF	Support in AG
1.	D0 – CVSD on SCO link (HV1)	M	M
2.	D1 – CVSD on SCO link (HV3)	M	M
3.	S1 – CVSD eSCO link (EV3)	M	M
4.	S2 – CVSD on EDR eSCO link (2-EV3)	M	M
5.	S3 – CVSD on EDR eSCO link (2-EV3)	M	M
6.	S4 – CVSD on EDR eSCO link (2-EV3)	M	M
7.	T1 – mSBC on eSCO link (EV3)	C1	C1
8.	T2 – mSBC on EDR eSCO link (2-EV3)	C1	C1
9.	T1 – LC3-SWB on eSCO link (EV3)	C2	C2
10.	T2 – LC3-SWB on EDR eSCO link (2-EV3)	C2	C2
C1: Mandatory if Wide Band Speech is supported, excluded otherwise C2: Mandatory if Super Wide Band Speech is supported, excluded otherwise			

Table 3.4: Codec to link feature mapping

4 Hands-Free Control Interoperability Requirements

4.1 Introduction

The interoperability requirements for the Hands-Free Control entity are completely contained in this section. Sections 4.2 through 4.29 specify the requirements for the procedures directly related to the application layer features.

The procedures listed in this section are primarily based on the use of a minimum set of AT commands as the control protocol. Section 5 specifies these AT commands and their result codes.

Section 4.2 specifies how Service Level Connections are handled in general and specifically states how the layers beneath the Hands-Free Control entity are used to establish and release a Service Level Connection.

4.2 Service Level Connection Establishment

Upon a user action or an internal event, either the HF or the AG may initiate a Service Level Connection establishment procedure.

A Service Level Connection establishment requires the existence of a RFCOMM connection, that is, a RFCOMM data link channel between the HF and the AG.

Both the HF and the AG may initiate the RFCOMM connection establishment. If there is no RFCOMM session between the AG and the HF, the initiating device shall first initialize RFCOMM.

The RFCOMM connection establishment shall be performed as described in Generic Access Profile (Volume 3, Part C, Section 7.3 in [1]) and Section 6.1.1.

4.2.1 Service Level Connection Initialization

When an RFCOMM connection has been established, the Service Level Connection Initialization procedure shall be executed.

4.2.1.1 Supported features exchange

First, in the initialization procedure, the HF shall send the AT+BRSF=<HF supported features> command to the AG to both notify the AG of the supported features in the HF, as well as to retrieve the supported features in the AG using the +BRSF result code.[1]

4.2.1.2 Codec Negotiation

Secondly, in the initialization procedure, if the HF supports the Codec Negotiation feature, it shall check if the AT+BRSF command response from the AG has indicated that it supports the Codec Negotiation feature. If both the HF and AG do support the Codec Negotiation feature then the HF shall send the AT+BAC=<HF available codecs> command to the AG to notify the AG of the available codecs in the HF.[2]

4.2.1.3 AG Indicators

After having retrieved the supported features in the AG, the HF shall determine which indicators are supported by the AG, as well as the ordering of the supported indicators. This is because, according to the 3GPP 27.007 specification [2], the AG may support additional indicators not provided for by the

[1] Audio Gateways supporting the 0.96 version of Hands-Free Profile will return ERROR as a response to AT+BRSF

[2] Legacy devices shall not indicate support for the Codec Negotiation feature

Hands-Free Profile, and because the ordering of the indicators is implementation specific. The HF uses the AT+CIND=? Test command to retrieve information about the supported indicators and their ordering.

Once the HF has the necessary supported indicator and ordering information, it shall retrieve the current status of the indicators in the AG using the AT+CIND? Read command.

After having retrieved the status of the indicators in the AG, the HF shall then enable the "Indicators status update" function in the AG by issuing the AT+CMER command, to which the AG shall respond with OK. As a result, the AG shall send the +CIEV unsolicited result code with the corresponding indicator value whenever a change in service, call, or call setup status occurs. When an update is required for both the call and call setup indicators, the AG shall send the +CIEV unsolicited result code for the call indicator before sending the +CIEV unsolicited result code for the call setup indicator. The HF shall use the information provided by the +CIEV code to update its own internal and/or external indications.

Once the "Indicators status update" function has been enabled, the AG shall keep the function enabled until either the AT+CMER command is issued to disable it, or the current Service Level Connection between the AG and the HF is dropped for any reason.

After the HF has enabled the “Indicators status update” function in the AG, and if the “Call waiting and 3-way calling” bit was set in the supported features bitmap by both the HF and the AG, the HF shall issue the AT+CHLD=? test command to retrieve the information about how the call hold and multiparty services are supported in the AG. The HF shall not issue the AT+CHLD=? test command in case either the HF or the AG does not support the "Three-way calling" feature.

4.2.1.4 HF Indicators

If the HF supports the HF indicator feature, it shall check the +BRSF response to see if the AG also supports the HF Indicator feature.

If both the HF and AG support the HF Indicator feature, then the HF shall send the AT+BIND=<HF supported HF indicators> command to the AG to notify the AG of the supported indicators’ assigned numbers in the HF. The AG shall respond with OK.

After having provided the AG with the HF indicators it supports, the HF shall send the AT+BIND=? to request HF indicators supported by the AG. The AG shall reply with the +BIND response listing all HF indicators that it supports followed by an OK.

Once the HF receives the supported HF indicators list from the AG, the HF shall send the AT+BIND? command to determine which HF indicators are enabled. The AG shall respond with one or more +BIND responses. The AG shall terminate the list with OK. (See Section 4.35.3).

From this point onwards, the HF may send the AT+BIEV command with the corresponding HF indicator value whenever a change in value occurs of an enabled HF indicator.

The AG may enable or disable the notification of any HF indicator at any time by using the +BIND unsolicited response (See Section 4.35.4).

4.2.1.5 Completion of Service Level Connection Initialization

The HF shall consider the Service Level Connection fully initialized, and thereby established, in either of the following cases:

- After the HF has successfully retrieved information about HF indicators currently enabled by the AG using the AT+BIND? command, if and only if the “HF indicators” bit was set in the HF supported features bitmap and the AG supported features bitmap as exchanged via +BRSF command.

- After the HF has successfully retrieved information about how call hold and multiparty services are supported in the AG using the AT+CHLD command, if and only if the “Call waiting and 3-way calling” bit was set in the SupportedFeatures attribute of the SDP records for both HF and AG. This case shall apply when the “HF Indicators” bit is not set in the supported features bitmap for either the HF or the AG as exchanged via +BRSF command.
- After the HF has successfully enabled the “Indicator status update” using the AT+CMER command. This case shall apply when the “Call waiting and 3-way calling” bit is not set in the SupportedFeatures attribute of the SDP records for either the HF or the AG, and when the “HF Indicators” bit is not set in the supported features bitmap for either the HF or the AG as exchanged via +BRSF command.

If the HF receives an indication from the AG that a call is currently active, the HF may determine if an unanswered call is currently on hold by querying the Response and Hold state of the AG (see Section 4.30.1).

The AG shall consider the Service Level Connection to be fully initialized, and thereby established, in either of the following cases:

- After the AG has successfully responded with information about which HF indicators are enabled on the AG using +BIND as well as responded OK, if and only if the “HF Indicators” bit was set in the HF supported features bitmap and the AG supported features bitmap as exchanged via +BRSF command.
- After that the AG has successfully responded with information about how call hold and multiparty services are supported in the AG using +CHLD as well as responded OK, if and only if the “Call waiting and 3-way calling” bit was set in the SupportedFeatures of the SDP attribute for both HF and AG. This case shall apply when the “HF Indicators” bit is not set in the supported features bitmap for either the HF or the AG as exchanged via +BRSF command.
- After the AG has successfully responded with OK to the AT+CMER command (to enable the “Indicator status update” function.) This case shall apply when the “Call waiting and 3-way calling” bit is not set in the supported features bitmap for either the HF or the AG, and when the “HF Indicators” bit is not set in the supported features bitmap for either the HF or the AG as exchanged via +BRSF command.

See Section 5 for more information on the AT+CIND, AT+CMER, AT+CHLD, AT+BAC and AT+BRSF commands and the +CIEV unsolicited result code.

4.2.1.6 Service Level Connection Diagram

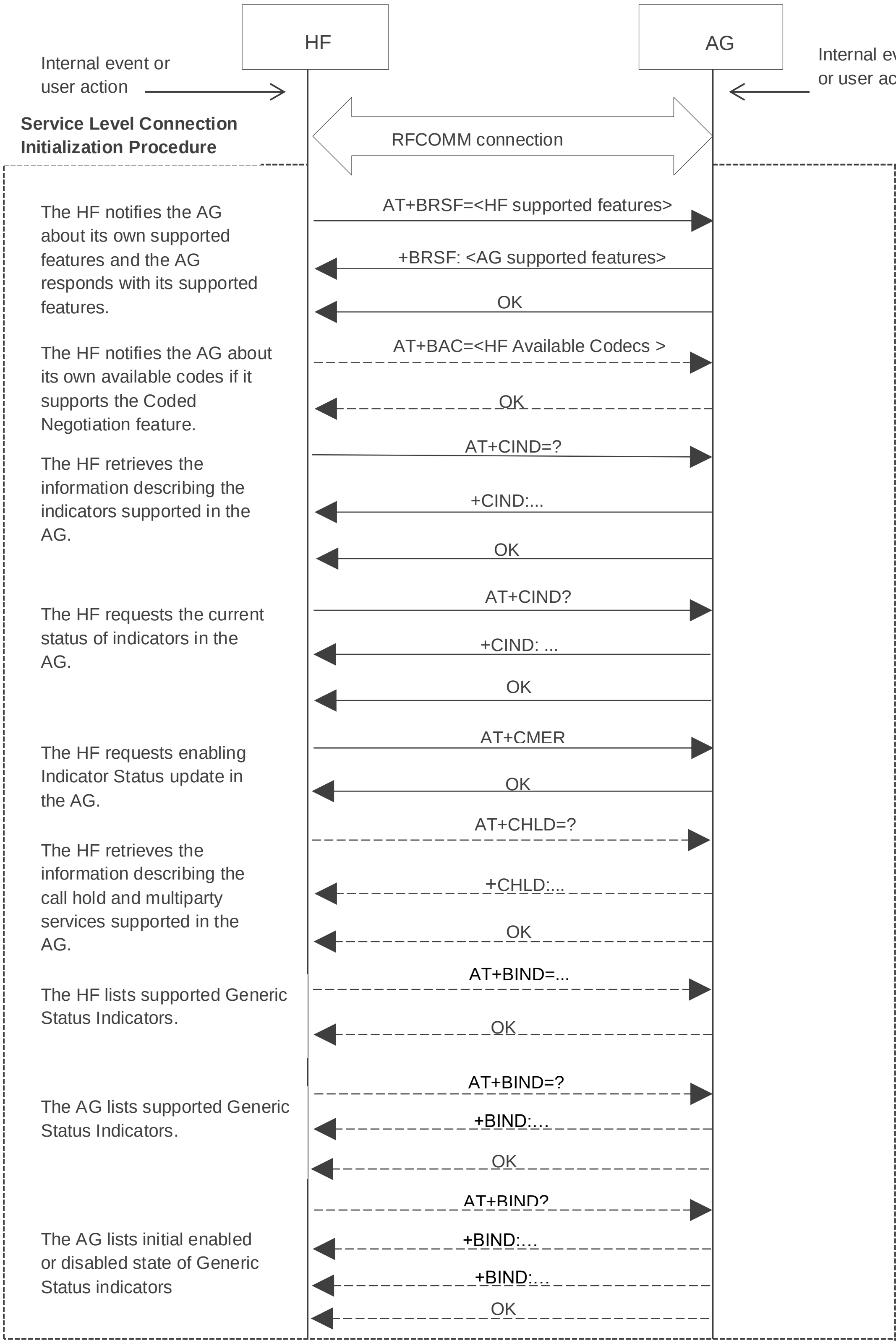

Figure 4.1: Service Level Connection establishment

4.2.2 Link Loss Recovery

This section addresses the link loss recovery from an HF. The HF may reconnect with the AG whenever there is loss of Bluetooth link.

When a Service Level Connection is disconnected due to explicit termination at one end (using the "Service connection release" as described in Section Service Level Connection Release), then both devices (AG and HF) shall wait for an explicit user action before an attempt is made to re-establish the Service Level Connection.

If the HF determines that the Service Level Connection was disconnected due to a link supervision timeout, then the HF may execute the "Service Level Connection establishment" procedure as described in Section 4.2 to establish a new Service Level Connection to the AG. Following a link loss due to link supervision timeout, the HF shall not assume that the service level connection state from the previous connection is valid (such as Call Status, Service Status).

4.3 Service Level Connection Release

This section describes the procedure for releasing a Service Level Connection.

The disconnection of a Service Level Connection shall result in the immediate removal of the corresponding RFCOMM data link channel between the HF and the AG. Also, an existing audio connection has to be removed as a consequence of the removal of the Service Level Connection. The removal of the L2CAP and link layers is optional.

An established Service Level Connection shall be released using a “Service Level Connection removal” procedure.

- Either the HF or AG shall initiate the “Service Level Connection release” procedure due to an explicit user request.
- The “Service Level Connection release” procedure shall be initiated if the Bluetooth functionality is disabled in either the HF or AG.
- The “Service Level Connection release” procedure may be initiated if an “Audio Connection transfer towards the AG”, as stated in Section 4.17, is performed during an ongoing call in the AG. In the case that the Service Level Connection is removed, the AG shall attempt to re-establish the Service Level Connection once the call is dropped.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist.

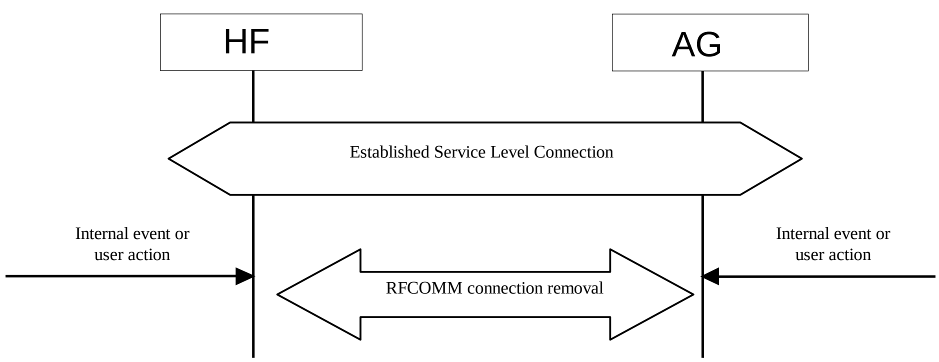

Figure 4.2: Service Level Connection removal

4.4 Transfer of Registration Status

The AT+CMER command, as described in Section 5, enables the "Registration status update" function in the AG.

The AT+BIA command, as described in Section 5.3, allows the HF to deactivate/reactivate individual indicators.

When the CMER function is enabled and the registration status indicator has not been deactivated by the AT+BIA command, the AG shall send the +CIEV unsolicited result code with the corresponding service indicator and value whenever the AG's registration status changes. The HF shall be capable of interpreting the information provided by the +CIEV result code to determine the service availability status as listed in Section 5.2.

If the CMER function is disabled or the indicator has been deactivated by the AT+BIA command, the AG shall not send the unsolicited result code.

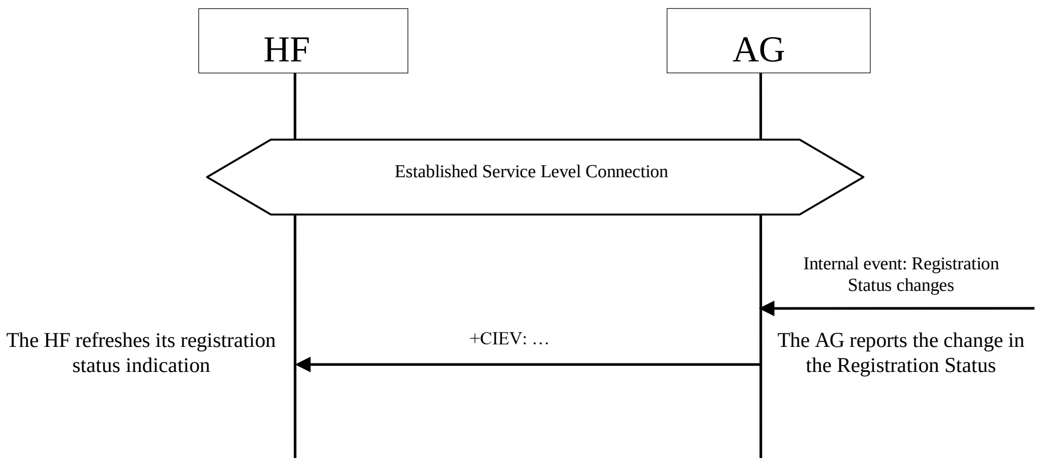

Figure 4.3: Typical Registration Status update

4.5 Transfer of Signal Strength Indication

The AT+CMER command, as described in Section 5, enables the "Signal strength Indication" function in the AG.

The AT+BIA command, as described in Section 5.3, allows the HF to deactivate/reactivate individual indicators.

When the CMER function is enabled and the indicator has not been deactivated by the AT+BIA command, the AG shall send the +CIEV unsolicited result code with the corresponding signal strength indicator and value whenever the AG's signal strength changes. The HF shall be capable of interpreting the information provided by the +CIEV result code to determine the signal strength as listed in Section 5.2.

If the CMER function is disabled or the indicator has been deactivated by the AT+BIA command, the AG shall not send the unsolicited result code.

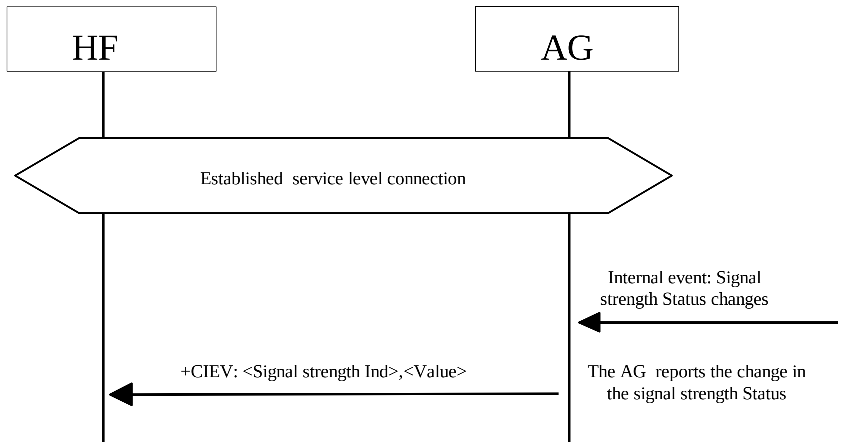

Figure 4.4: Transfer of Signal strength indication

As a result, the AG shall send the +CIEV unsolicited result code with the corresponding signal strength value whenever its signal strength changes.

4.6 Transfer of Roaming Status Indication

The AT+CMER command, as described in Section 5, enables the “Roaming Status Indication” function in the AG.

The AT+BIA command, as described in Section 5.3, allows the HF to deactivate/reactivate individual indicators.

When the CMER function is enabled and the indicator has not been deactivated by the AT+BIA command, the AG shall send the +CIEV unsolicited result code with the corresponding roaming status indicator and value whenever the AG's roaming status changes. The HF shall be capable of interpreting the information provided by the +CIEV result code to determine the roaming status as listed in Section 5.2.

If the CMER function is disabled or the indicator has been deactivated by the AT+BIA command, the AG shall not send the unsolicited result code.

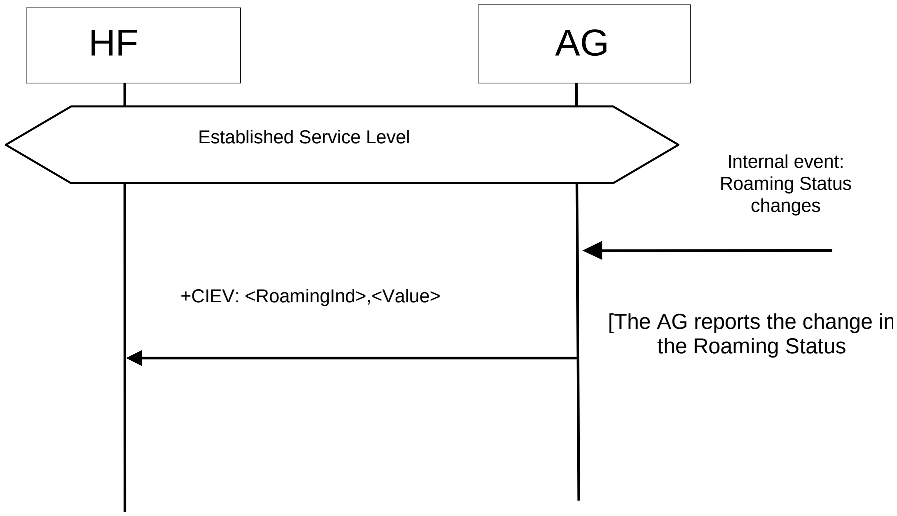

Figure 4.5: Transfer of Roaming Status Indication

4.7 Transfer of Battery Level Indication of AG

The AT+CMER command, as described in Section 5 enables the "Battery Level Indication" function in the AG.

The AT+BIA command, as described in Section 5.3, allows the HF to deactivate/reactivate individual indicators.

When the CMER function is enabled and the indicator has not been deactivated by the AT+BIA command, the AG shall send the +CIEV unsolicited result code with the corresponding battery level indicator and value whenever the AG's battery level changes. The HF shall be capable of interpreting the information provided by the +CIEV result code to determine the battery level as listed in Section 5.2.

If the CMER function is disabled or the indicator has been deactivated by the AT+BIA command, the AG shall not send the unsolicited result code.

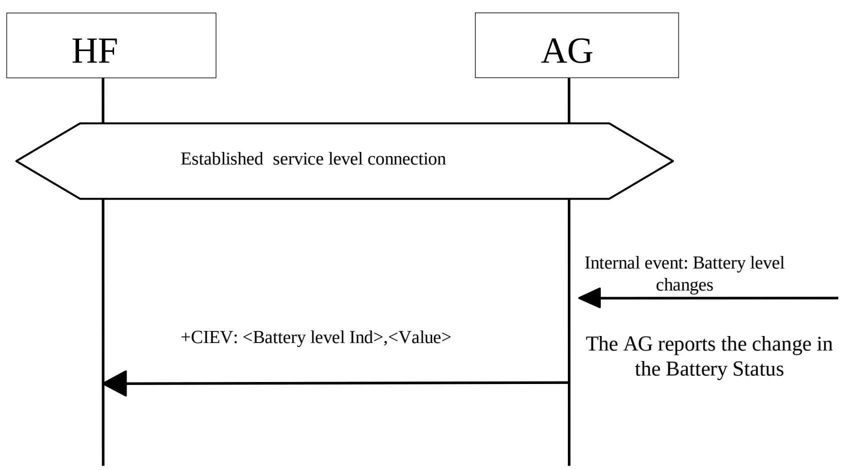

Figure 4.6: Transfer of Battery level indication

4.8 Query Operator Selection

The HF shall execute this procedure to find out the name of the currently selected Network operator by AG.

The following precondition applies for this procedure:

- An ongoing connection between the AG and the HF shall exist. If this connection did not exist, the AG shall establish a connection using “Service Level Connection set up” procedure described in Section 4.2.

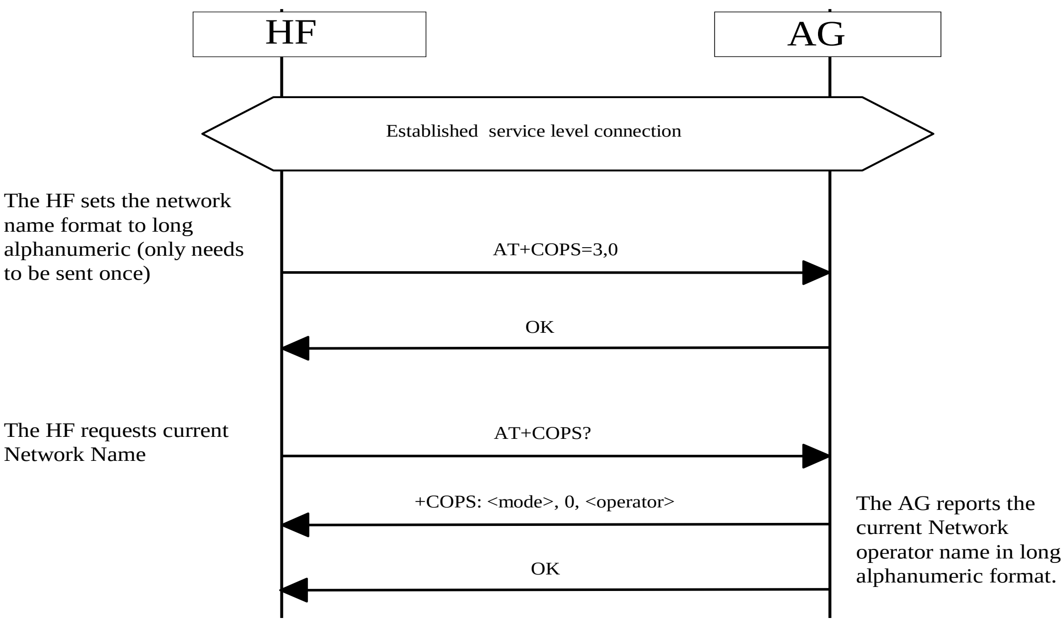

Figure 4.7: Query currently selected Network operator

- HF shall send AT+COPS=3,0 command to set name format to long alphanumeric. Long alphanumeric is defined as a maximum of 16 characters. The value of 3 as the first parameter indicates that this command is only affecting the format parameter (the second parameter). The second parameter, 0, is the value required to set the format to “long alphanumeric.”
- Upon an internal event or user-initiated action, HF shall send the AT+COPS? (Read) command to find the currently selected operator.
- AG shall respond with +COPS response indicating the currently selected operator. If no operator is selected, <format> and <operator> are omitted.

4.9 Report Extended Audio Gateway Error Results Code

The HF shall execute this procedure to enable/disable Extended Audio Gateway Error result codes in the AG.

The following precondition applies for this procedure:

- An ongoing connection between the AG and the HF shall exist. If this connection did not exist, the AG shall establish a connection using "Service Level Connection set up" procedure described in Section 4.2.

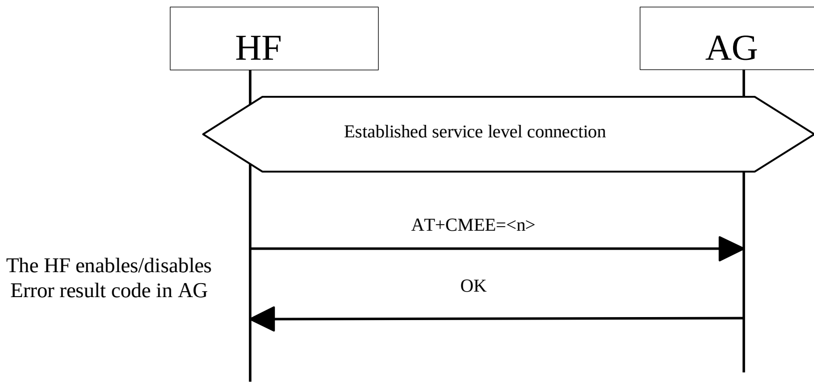

Figure 4.8: Enable/Disable AG Error result code

- The HF shall issue the AT+CMEE command to enable/disable the "Extended Audio Gateway Error Result Code" in the AG. The parameter <n> controls the activation/deactivation of the "Extended Error result code" notification.
- Whenever there is an error relating to the functionality of the AG as a result of AT command, the AG shall send +CME ERROR: <err> response to the HF.

4.10 Transfer of Call, Call Setup, and Held Call Status

The AT+CMER command, as described in Section 5, enables the "Call Status indicators update" function in the AG.

The AT+BIA command, as described in Section 5.3, allows the HF to deactivate/reactivate individual indicators. The AT+BIA command shall have no effect on the Call, Call Setup, or Held Call indicators. These indicators cannot be deactivated using the AT+BIA command.

When the CMER function is enabled, the AG shall issue a +CIEV unsolicited result code with the corresponding call indicator and value whenever the AG's current call status changes. Likewise, the AG shall issue a +CIEV unsolicited result code with the corresponding callsetup indicator and value whenever the AG's current call setup status changes. The AG shall also issue a +CIEV unsolicited result code with corresponding callheld indicator and value whenever the AG's current held call status changes.

If the CMER function is disabled, the AG shall not send the unsolicited result code.

The HF shall be capable of interpreting the information provided by the +CIEV result code to determine the call status as listed in Section 5.2.

Furthermore, the HF may also be capable of interpreting the optional callsetup state information provided by the +CIEV result code as listed in Section 5.2.

The HF shall be able to accept unknown indicators provided by the +CIEV result code. The HF may ignore unknown indicators provided by the +CIEV result code.

Note: Although the HF is required to parse the +CIEV result codes, the HF is not required to provide User Interface indicators for the +CIEV result codes.

4.10.1 Transfer of Call Status

Upon the change in status of any call in the AG, the AG shall execute this procedure to advise the HF of the current call status changes. The values for the call indicator are:

0= No calls (held or active)
1= Call is present (active or held)

The following precondition applies for this procedure:

- The HF shall have enabled the "Indicators status update" in the AG.
- A Service Level Connection shall exist between the AG and HF devices.

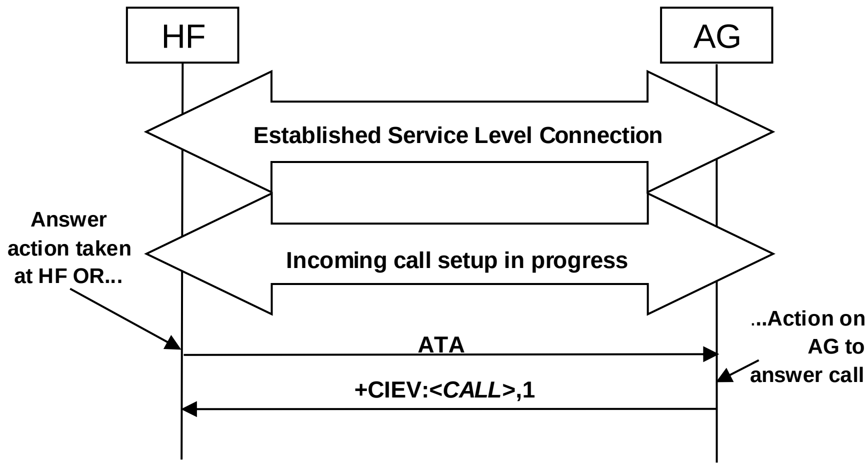

Figure 4.9: Call Status: Incoming call answered

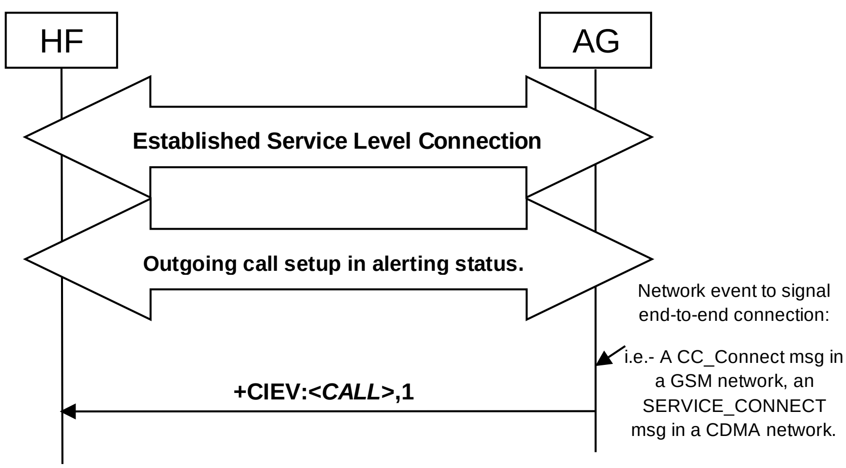

Figure 4.10: Call Status: Outgoing call

Consequently, upon the release of any call that places the AG in a state where there is no call present on the AG, either by the AG, a network event, or actions by the HF, the AG shall issue a +CIEV unsolicited result code with the call indicator value of “0”.

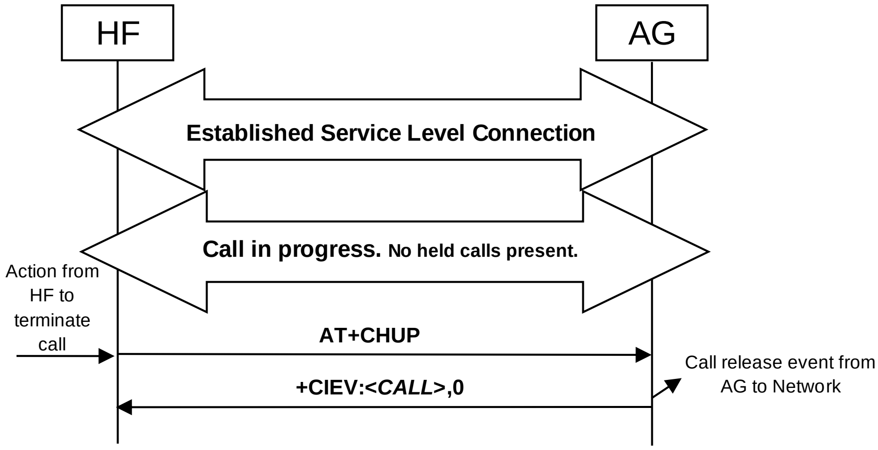

Figure 4.11: Call Status: Call Release from HF

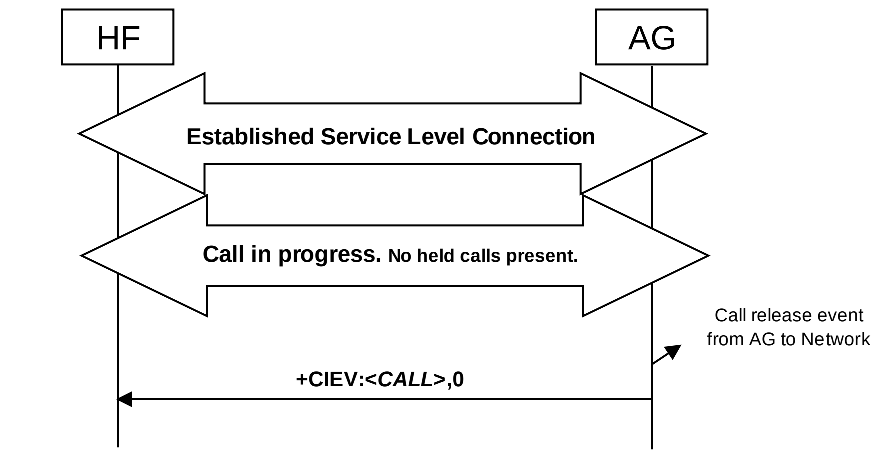

Figure 4.12: Call Status: Call Release from AG

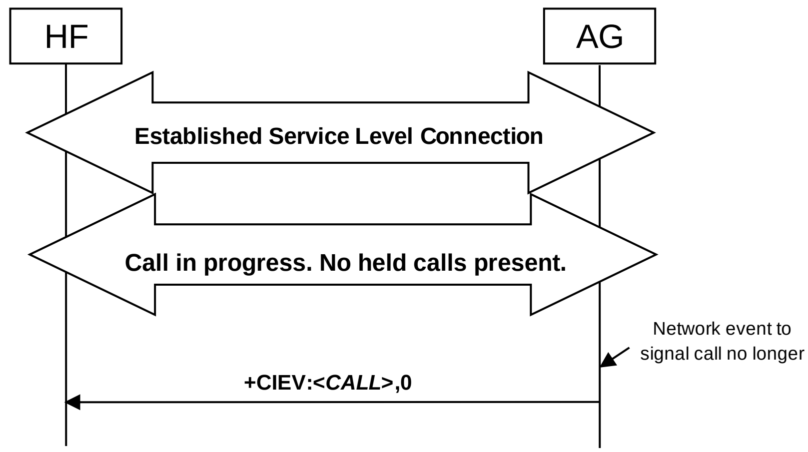

Figure 4.13: Call Status: Call Release from Network

4.10.2 Transfer of Callsetup Status

Upon the change in AG's callsetup status, the AG shall execute this procedure to advise the HF of the current callsetup status changes. The values for the callsetup indicator are:

0= No call setup in progress.

1= Incoming call setup in progress.

2= Outgoing call setup in dialing state.

3= Outgoing call setup in alerting state.

The following precondition applies for this procedure:

- The HF shall have enabled the "Indicators status update" in the AG.
- A Service Level Connection shall exist between the AG and HF devices.

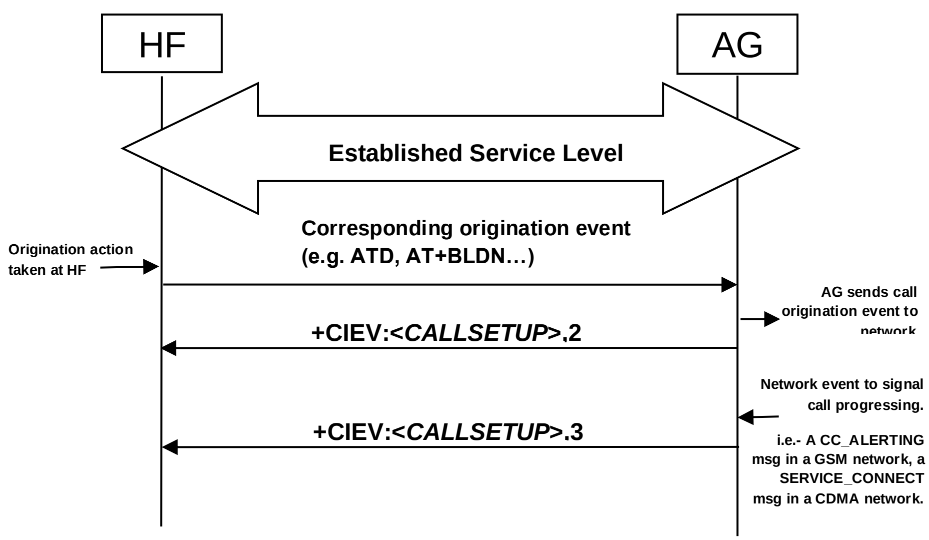

Figure 4.14: Callsetup Status: Outgoing

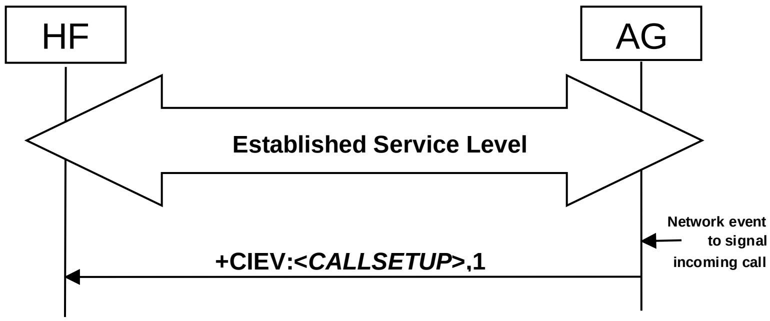

Figure 4.15: Callsetup Status: Incoming (See Section 4.13, 4.14)

Once the AG has been advised via whatever applicable network function, that a call has reached an end-to-end connected status, the callsetup indicator shall be reset indicating that no call setup procedures are in progress.

4.10.3 Indication of Status for Held Calls

Upon the change in status of any call on hold in the AG, the AG shall execute this procedure to advise the HF of the held call status. The values for the callheld indicator are:

0= No calls held

1= Call is placed on hold or active/held calls swapped

(The AG has both an active AND a held call)

2= Call on hold, no active call

The following precondition applies for this procedure:

- The HF shall have enabled the Call Status Indicators function in the AG.
- A Service Level Connection shall exist between the AG and HF devices.

Whenever an active call is placed on hold such that the AG now has both an active and held call or the active/held call positions swapped by a request from the HF or by action on the AG, the AG shall issue a +CIEV unsolicited result code with the callheld indicator value of “1”.

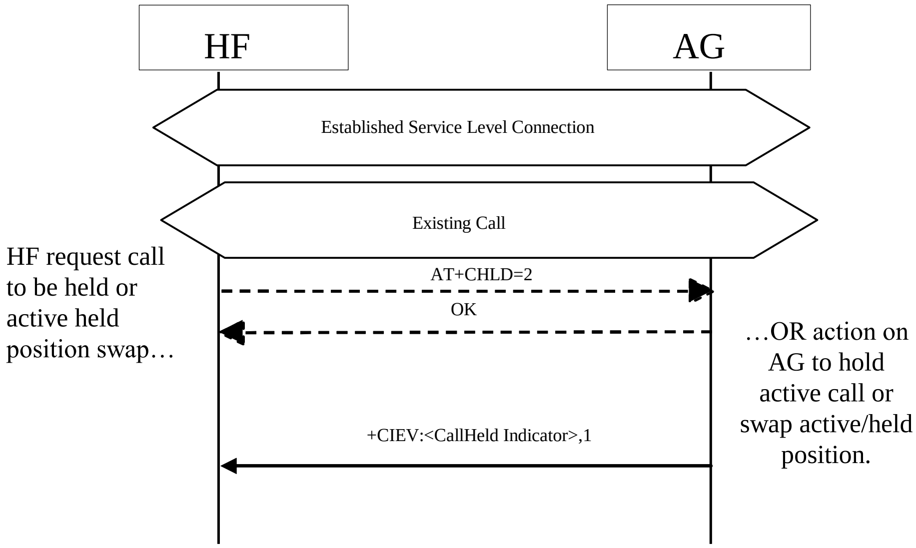

Figure 4.16: Call Held or Active/Held Position Swap

Consequently, upon the release of any call on hold by the HF, the AG or by network event, or actions by the HF or AG to retrieve a held call, the AG shall issue a +CIEV unsolicited result code with the callheld indicator value of “0”.

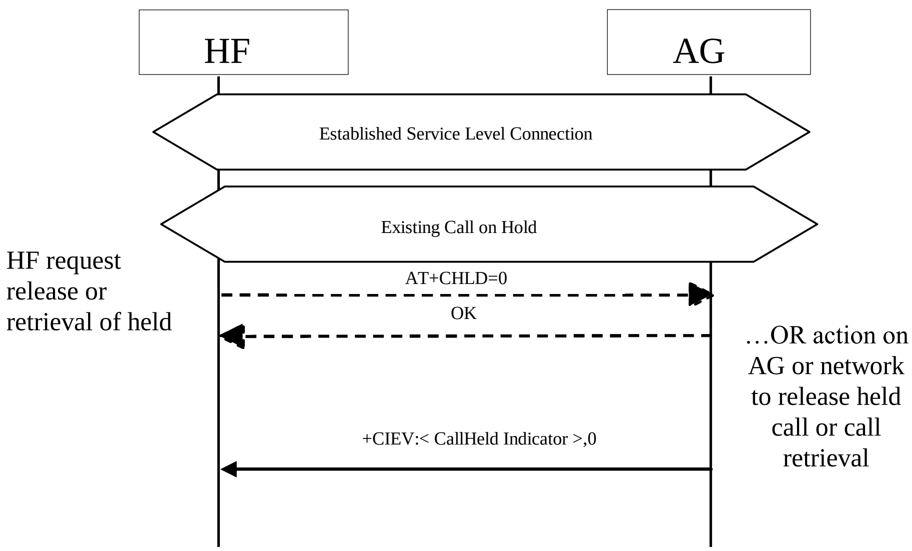

Figure 4.17: Held Call Release

If a call is still on hold when an active call is terminated or a single active call is put on hold, the AG shall issue a +CIEV unsolicited result code with the callheld indicator value of “2”.

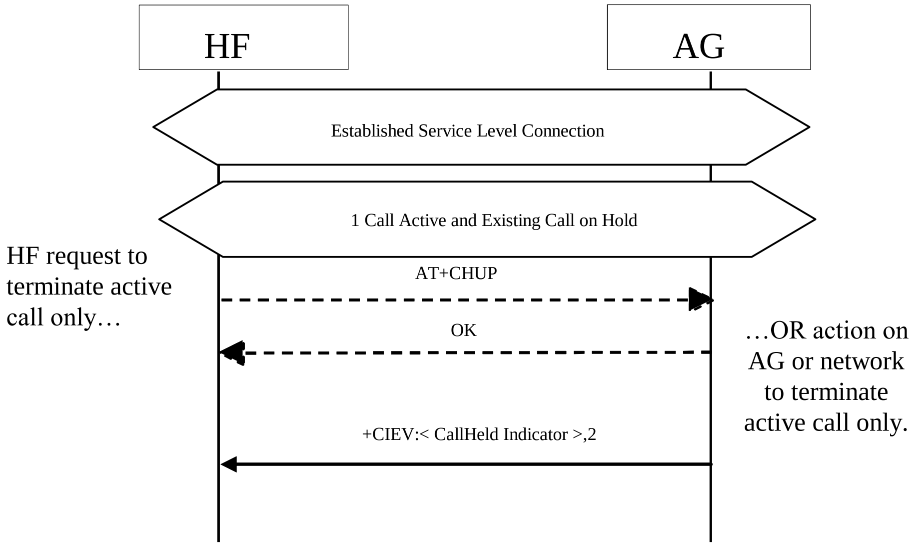

Figure 4.18: Active Call Terminated/Call Remains Held

4.11 Audio Connection Setup

Upon user action or an internal event, either the HF or the AG may initiate the establishment of an audio connection as needed. Further internal actions may be needed by the HF or the AG to internally route, modify sample rate, frame and/or sample alignment of the audio paths. More formally stated, the requirements for audio connection setup are:

- The HF shall be capable of initiating an audio connection during a call process.
- The HF may be capable of initiating an audio connection while no call is in process.
- The AG shall be capable of initiating an audio connection during a call process.
- The AG may be capable of initiating an audio connection while no call is in process.

Incoming call audio handling is further described in Section 4.12.

Outgoing call audio handling is further described in Sections 4.17, 4.18, and 4.19.

An Audio Connection set up procedure always means the establishment of a Synchronous Connection and it is always associated with an existing Service Level Connection.

In principle, setting up an Audio Connection by using the procedure described in this section is not necessarily related to any call process.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the initiator of the procedure shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

Both the initiator and the acceptor shall notify the presence of the new Audio Connection.

4.11.1 Audio Connection Setup by AG

When the AG is to setup an Audio Connection, it shall initiate the Codec Connection establishment procedure if the Service Level Negotiation indicated that both sides support this feature.

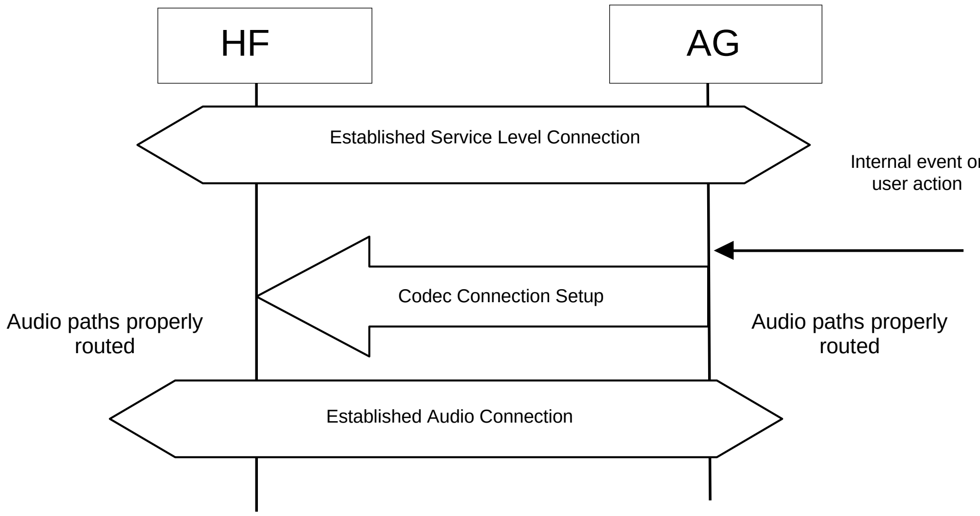

Figure 4.19: Procedure for Establishment of Audio Connection from AG

4.11.2 Audio Connection Setup by HF

For all HF initiated audio connection establishments for which both sides support the Codec Negotiation feature, the HF shall trigger the AG to establish a Codec Connection. This is necessary because only the AG knows about the codec selection and settings of the network.

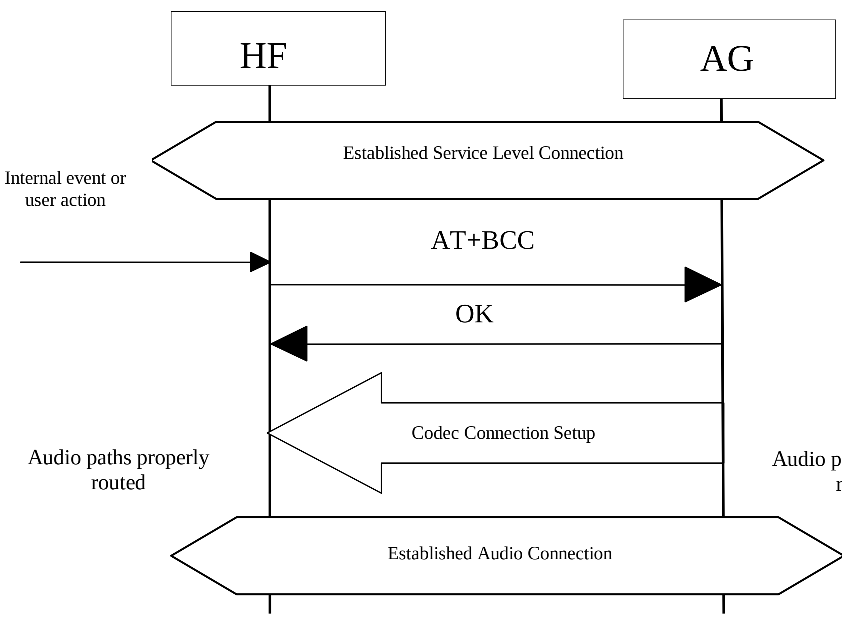

Figure 4.20: Procedure for Establishment of Audio Connection from HF

When the HF triggers the establishment of the Codec Connection it shall send the AT command AT+BCC to the AG. The AG shall respond with OK if it will start the Codec Connection procedure, and with ERROR if it cannot start the Codec Connection procedure.

After the AG has sent the OK response, the AG shall initiate the Codec Connection Setup procedure.

The type of Synchronous Connection that is established in this procedure and the settings used for it are dependent on the format of the media that is going to be transported over the connection and can be found in the Sections 4.2.1 and 6.7.2.

4.11.3 Codec Connection Setup

Upon user action or an internal event, either the HF or the AG may initiate the establishment of a Codec Connection Setup procedure, whenever necessary. Further internal actions may be needed by the HF or the AG to internally route, modify sample rate, frame and/or sample alignment of the audio paths.

Although the Audio Connection may be triggered by either the AG or the HF, the Codec Connection and the Synchronous Connection shall always be established by the AG (unless one of the devices is a legacy device).

The AG shall initiate a Codec Connection only if the HF has indicated support for the Codec Negotiation feature and has sent at least one AT+BAC on the current service level connection. When selecting which codec to use for a codec connection, the AG shall use the information on codecs available in the HF as provided in the most recently received AT+BAC.

The AG shall inform the HF which codec ID will be used before establishing the Synchronous Connection. If a codec has been successfully selected at least once on the current service level connection, the AG does not need to inform the HF about which codec to use unless a change of codec is required for the next synchronous connection. If the HF has sent an additional AT+BAC after the completion of the Service Level Connection procedure, there may be a need to re-select the codec.

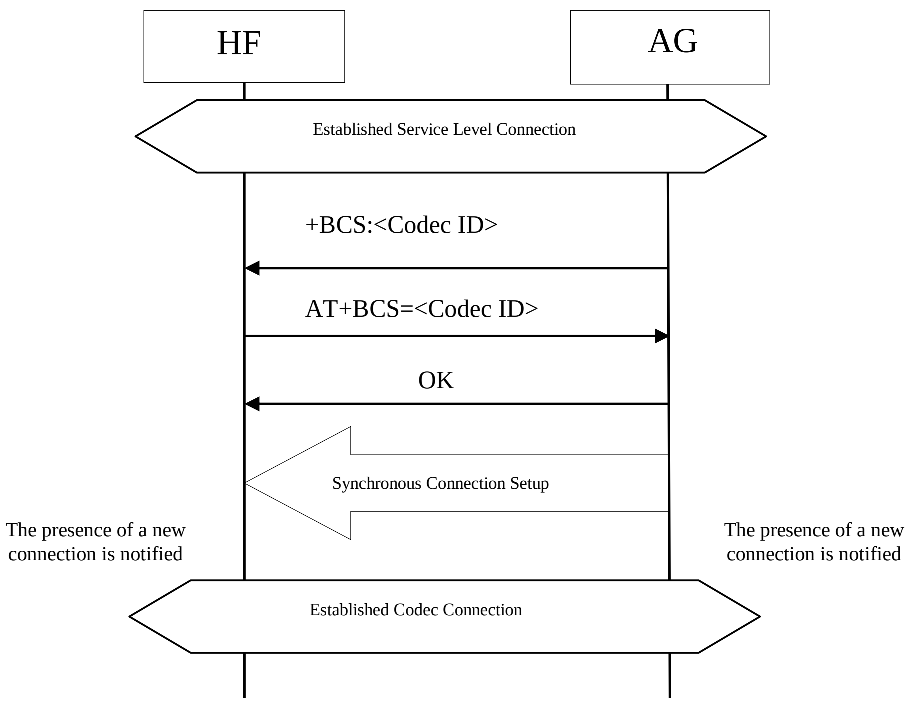

Figure 4.21: Procedure for the Establishment of Codec Connection

The AG shall send a +BCS=<Codec ID> unsolicited response to the HF. The HF shall then respond to the incoming unsolicited response with the AT command AT+BCS=<Codec ID>. The ID shall be the same as in the unsolicited response code as long as the ID is supported. If the received ID is not available, the HF shall respond with AT+BAC with its available codecs.

The AG shall always respond with OK if the ID received in AT+BCS is the same as the one sent, and with ERROR otherwise. If no AT+BCS is received, but instead an AT+BAC is received after sending +BCS, the procedure shall end but may be restarted by the AG after re-selecting codec ID based on the information in the just received AT+BAC.

After sending the OK response, the AG shall open the Synchronous Connection with the settings that are determined by the ID. The HF shall be ready to accept the synchronous connection establishment as soon as it has sent the AT commands AT+BCS=<Codec ID>.

After the Synchronous Connection has been established, the Codec Connection is established. The selection of codec with the +BCS command is in effect for this connection as well as subsequent Codec Connections.

If the Codec Connection establishment procedure fails before a Synchronous Connection has been established, the Codec Connection establishment procedure shall be re-started before any Synchronous Connection is established.

If an (e)SCO link cannot be established according to the parameters required for the selected codec (e.g., basebands negotiation fails for a link with re-transmission although a wide band or super wide band codec has been selected), the Codec Connection establishment procedure shall be re-started by the AG with the purpose of selecting a codec that can be used. The mandatory narrow band codec (CVSD) shall be used before the AG gives up trying to establish a Codec Connection.

The type settings for the Synchronous Connection that is established in this procedure are dependent on the format of the media that is going to be transported over the audio connection and can be found in the Section 6.7.2.

4.11.4 Available codecs updating

Any time on an established service level connection for which both sides support the Codec Negotiation Feature, the HF may send AT+BAC to inform the AG about dynamic changes in the set of available codecs, which does not mandate the closing of any existing audio connection. If the AG has started the Codec Connection Setup procedure, AT+BAC shall be sent by the HF in response to the +BCS unsolicited response from the AG when the selected codec has become unavailable.

If the last selected codec becomes unavailable, the HF shall send AT+BAC to the AG to prompt the AG to re-select a codec before the next Codec Connection set-up and AG shall send the +BCS unsolicited response before starting establishment of a Synchronous Connection.

The mandatory narrow band codec shall always be listed in the AT+BAC command. Hence, the AT+BAC shall never be an empty list; this provides a fallback option that is always available to setup an Audio Connection.

If the mandatory wide band codec or the mandatory super wide band codec is supported, it shall also always be listed in the AT+BAC commands unless support for Wide Band Speech or Super Wide Band Speech has become temporarily unavailable. If the HF has previously indicated in its AT+BAC command that it supports Wide Band Speech or Super Wide Band Speech, then the AG shall interpret this as temporary suspension of Wide Band Speech or Super Wide Band Speech support respectively until the HF sends the next AT+BAC command. If the mandatory Wide Band Speech codec is not included in the AT+BAC command, then no other Wide Band Speech codec shall be included. If the mandatory Super

Wide Band Speech codec is not included in the AT+BAC command, then no other Super Wide Band Speech codec shall be included.

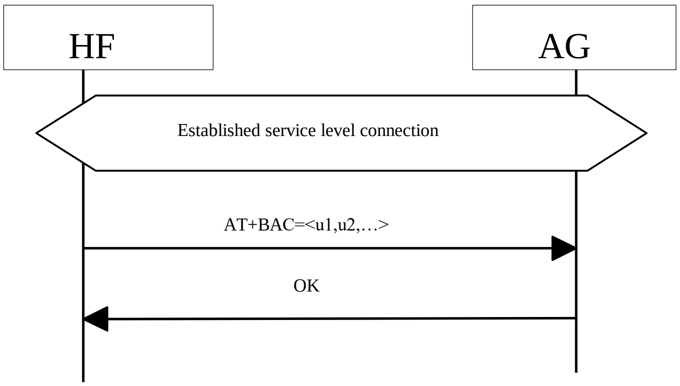

Figure 4.22: Procedure for updating the list of available codecs

4.11.5 Codec re-negotiation

When the AG establishes an Audio Connection, it will decide what codec to use based upon the list of available codecs communicated during the most recent AT+BAC command exchange. The selected Bluetooth codec shall then be used throughout the ongoing Synchronous Connection irrespective of any changes on the connection at the AG or HF side. To change the selected Bluetooth codec, the AG may initiate a Codec Connection Setup procedure. The newly selected codec as a result of this codec re-negotiation shall be used for the next Audio Connection.

4.12 Audio Connection Release

Upon a user action or an internal event, either the HF or the AG may release an existing Audio Connection. More formally stated, the requirements for audio connection release are:

- The HF shall be capable of releasing an audio connection during a call process.
- If the HF has the ability to set up an audio connection with no call in process, the HF shall be capable of releasing the audio connection while no call is in process.
- The AG shall be capable of releasing an audio connection during a call process.
- If the AG has the ability to set up an audio connection with no call in process, the AG shall be capable of releasing the audio connection while no call is in process.

As a precondition for this procedure, an ongoing Audio Connection between the AG and the HF shall exist.

An Audio Connection release always means the disconnection of its corresponding Synchronous Connection.

When the audio connection is released, the audio path shall be routed to the AG.[3]

[3] In principle, removing an Audio Connection by using the procedure described in this section is not necessarily related to any call process

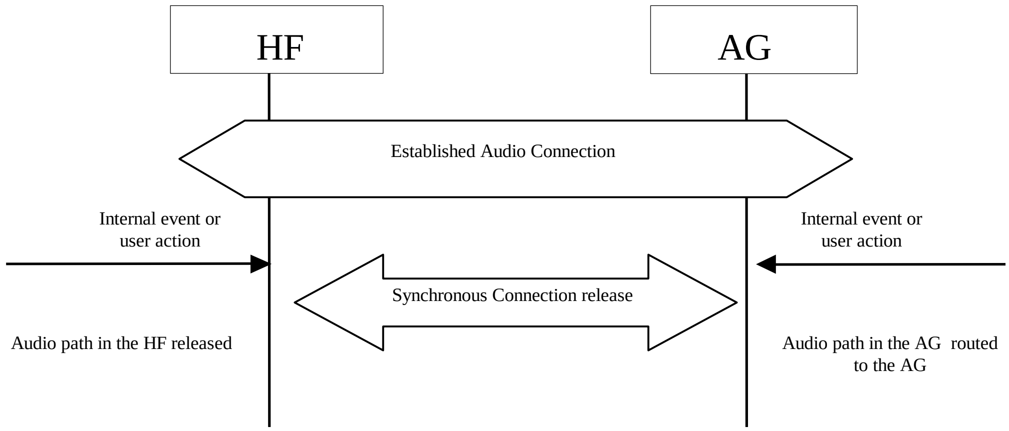

Figure 4.23: Audio Connection release

4.13 Answer an Incoming Call

Upon an incoming call, the AG shall send a sequence of unsolicited RING alerts to the HF. The RING alert shall be repeated for as long as the call acceptance is pending, or until the incoming call is interrupted for any reason.

The HF shall produce a local alerting in reaction to the RING.

If the AG's SDP record (or +BRSF message) indicates "In-band ring tone" is supported, the AG shall send in-band ring tones unless subsequently changed using procedures defined in Section 4.13.4.

The AG may abort the incoming call when necessary. It shall then stop sending the RING alert to the HF.

4.13.1 Answer Incoming Call from the HF – In-Band Ringing

Optionally, the AG may provide an in-band ring tone.

This case is described in Figure 4.24 below and implies as a precondition that an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the AG shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

As the figure below shows, if an in-band ring tone is used, the AG shall send the ring tone to the HF via the established Audio Connection.

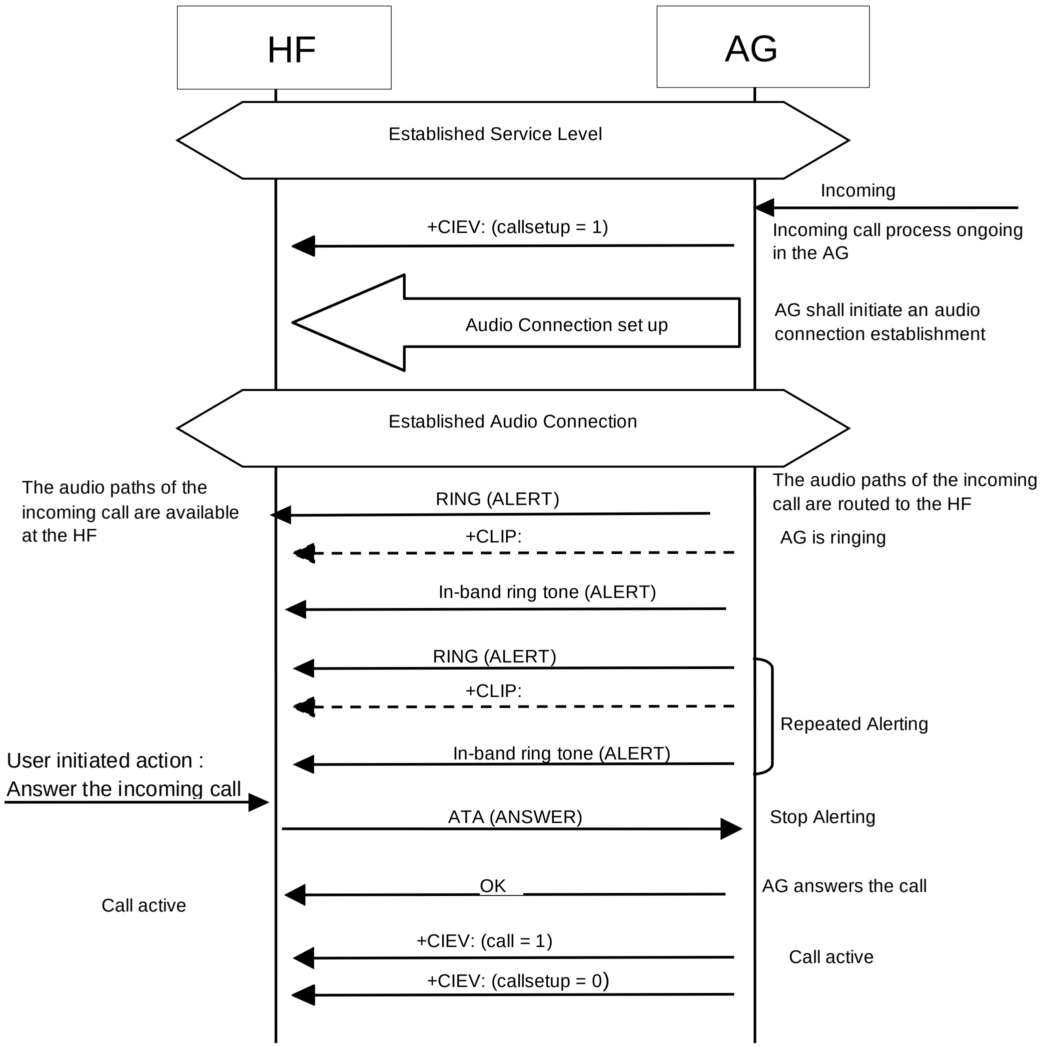

Figure 4.24: Answer an incoming call from the HF – in-band ring tone

The user accepts the incoming voice call by using the proper means provided by the HF. The HF shall then send the ATA command (see Section 5) to the AG. The AG shall then begin the procedure for accepting the incoming call.

If the normal incoming call procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0) to notify the HF of this condition (see also Section 4.14.2).

4.13.2 Answer Incoming Call from the HF – No In-Band Ringing

As a precondition, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the AG shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

As the figure below shows, if no in-band ring tone is used and an Audio Connection does not exist, the AG shall set up the Audio Connection and route the audio paths to the HF upon answering the call.

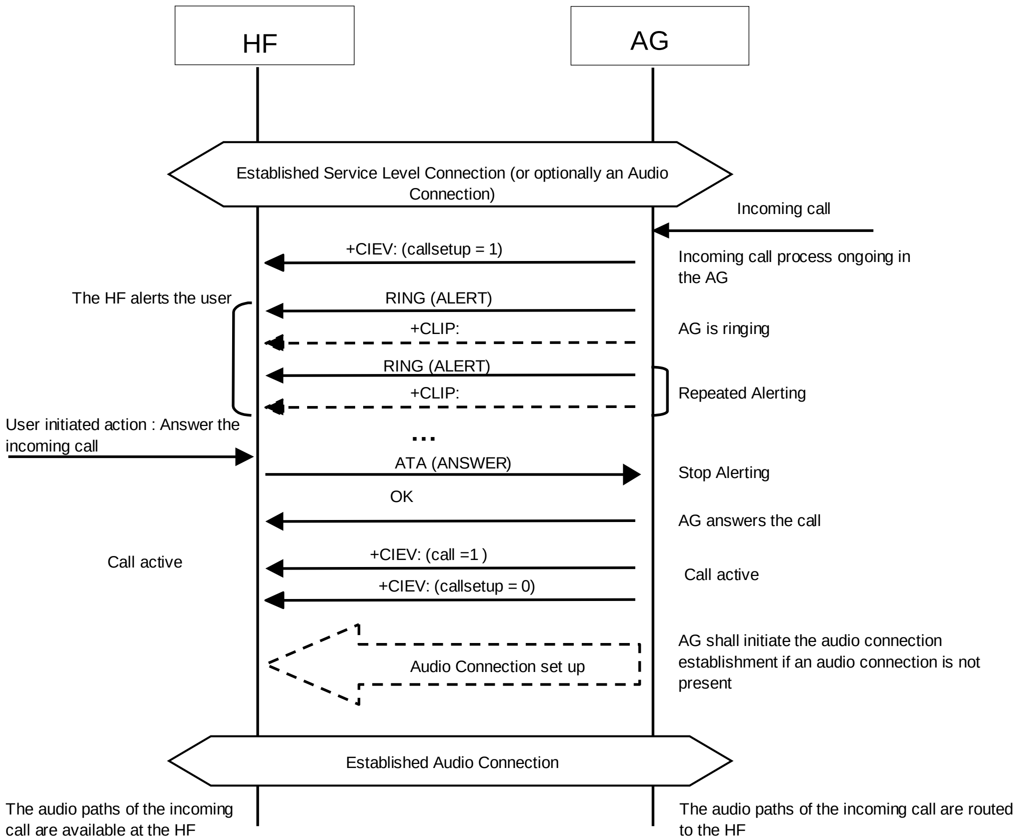

Figure 4.25: Answer an incoming call from the HF – no in-band ring tone

The user accepts the incoming voice call by using the proper means provided by the HF. The HF shall then send the ATA command (see Section 5) to the AG, and the AG shall start the procedure for accepting the incoming call and establishing the Audio Connection if an Audio Connection does not exist (refer to Section 4.10.1).

If the normal incoming call procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0) to notify the HF of this condition (see also Section 4.14.2).

4.13.3 Answer Incoming Call from the AG

The following preconditions apply for this procedure:

- As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist.
- The AG shall alert the HF using either of the two procedures described in Sections 4.13.1 and 4.13.2.
- The HF shall alert the user.

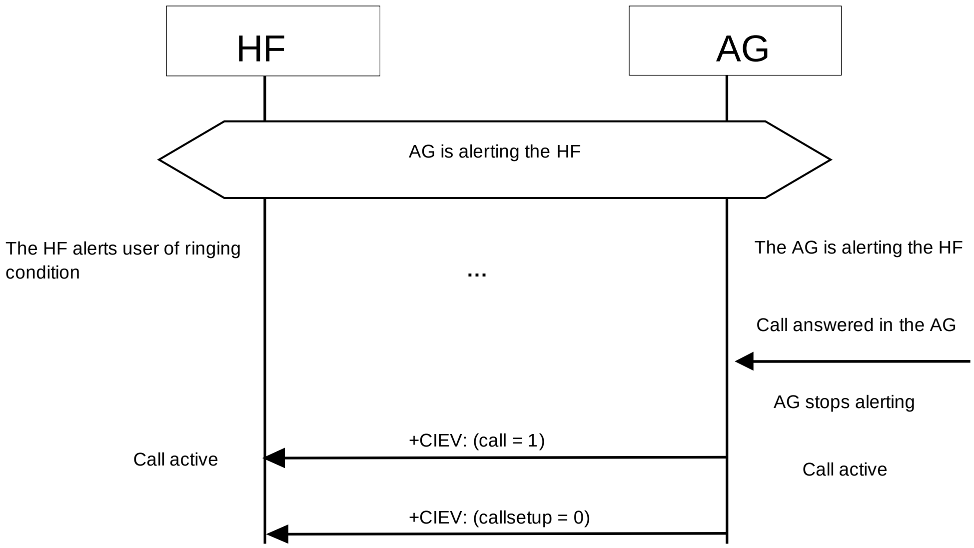

Figure 4.26: Answer an incoming call from the AG

The user accepts the incoming call by using the proper means provided by the AG.

If the normal incoming call procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0) to notify the HF of this condition (see also Section 4.14.2).

4.13.4 Change the In-Band Ring Tone Setting

The SDP record entry “In-band ring tone” of the “SupportedFeatures” record (see Table 6.6) informs the HF if the AG is capable of sending an in-band ring tone or not. If the AG is capable of sending an in-band ring tone, it shall send the in-band ring tone by default. The AG may subsequently change this setting.

If the AG wants to change the in-band ring tone setting during an ongoing service level connection, it shall use the unsolicited result code +BSIR (Bluetooth Set In-band Ring tone) to notify the HF about the change. See Figure 4.27 for details.

See Section 5 for more information on the +BSIR unsolicited result code.

The in-band ring tone setting may be changed several times during a Service Level Connection.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the AG shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

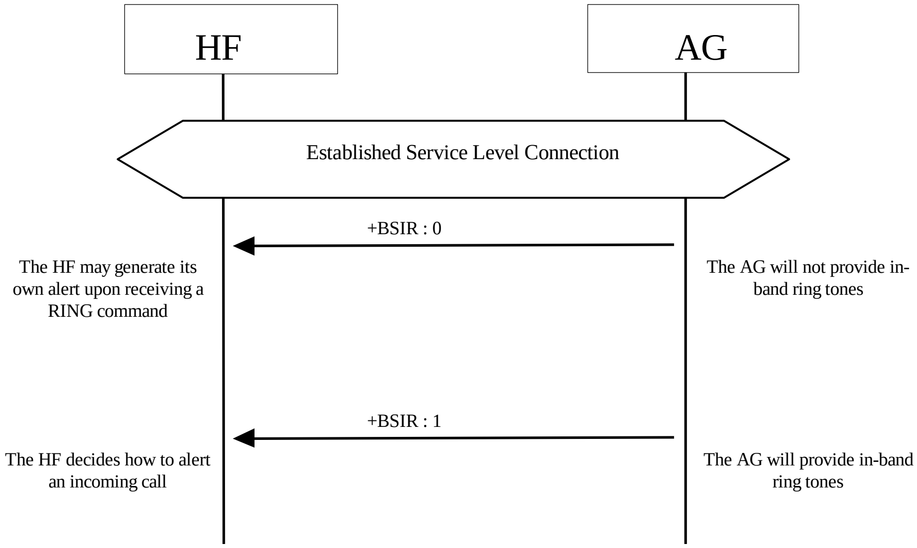

Figure 4.27: Change of the in-band ring tone setting initiated by the AG

If the HF does not want to use the AG's in-band ring tone, it may mute the Audio Connection after it has received +CIEV:(callsetup=1). The HF shall un-mute the Audio Connection upon receiving the +CIEV:(callsetup=0) indication.

4.14 Reject an Incoming Call

In case of an incoming call, the AG shall alert the HF by either one of the two procedures described in Sections 4.13.1 and 4.13.2.

Instead of answering the call, the user may reject the incoming call process by user action at the HF or the AG. These two procedures are described in the following sections.

4.14.1 Reject an Incoming Call from the HF

As a precondition to this procedure, the AG shall alert the HF using either of the two procedures described in Sections 4.13.1 and 4.13.2.

The user rejects the incoming call by using the User Interface on the Hands-Free unit. The HF shall then send the AT+CHUP command (see Section 5) to the AG. This may happen at any time during the procedures described in Sections 4.13.1 and 4.13.2.

The AG shall then cease alerting the HF of the incoming call and send the OK indication followed by the +CIEV result code, with the value indicating (callsetup=0).

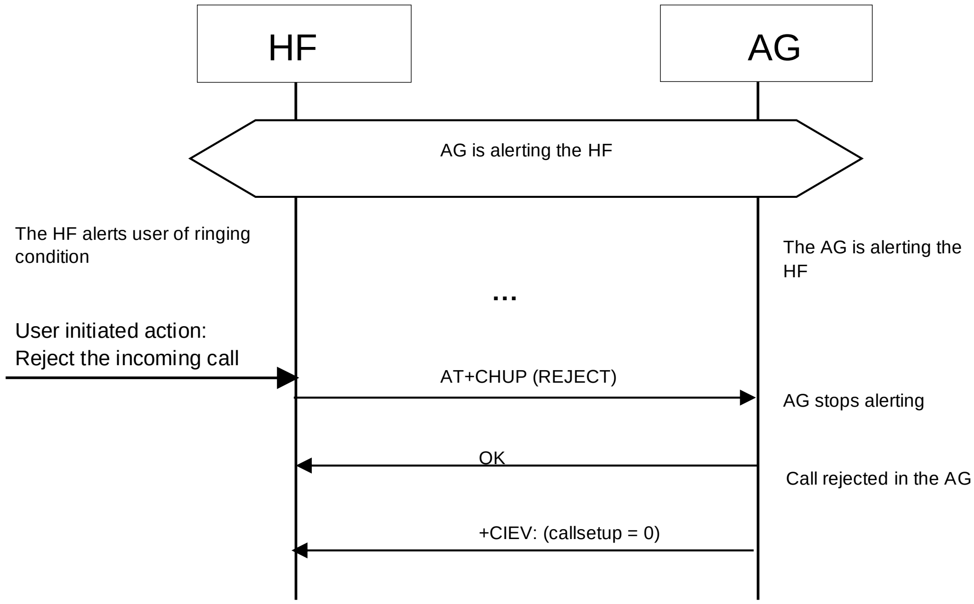

Figure 4.28: Reject an incoming call from the HF

4.14.2 Rejection/Interruption of an Incoming Call in the AG

As a precondition to this procedure, the AG shall alert the HF using either of the two procedures described in Sections 4.13.1 and 4.13.2.

The user rejects the incoming call by using the User Interface on the AG. Alternatively, the incoming call process may be interrupted in the AG for any other reason.

As a consequence of this, the AG shall send the +CIEV result code, with the value indicating (callsetup=0). The HF shall then stop alerting the user.

This may happen at any time during the procedures described in Sections 4.13.1 and 4.13.2.

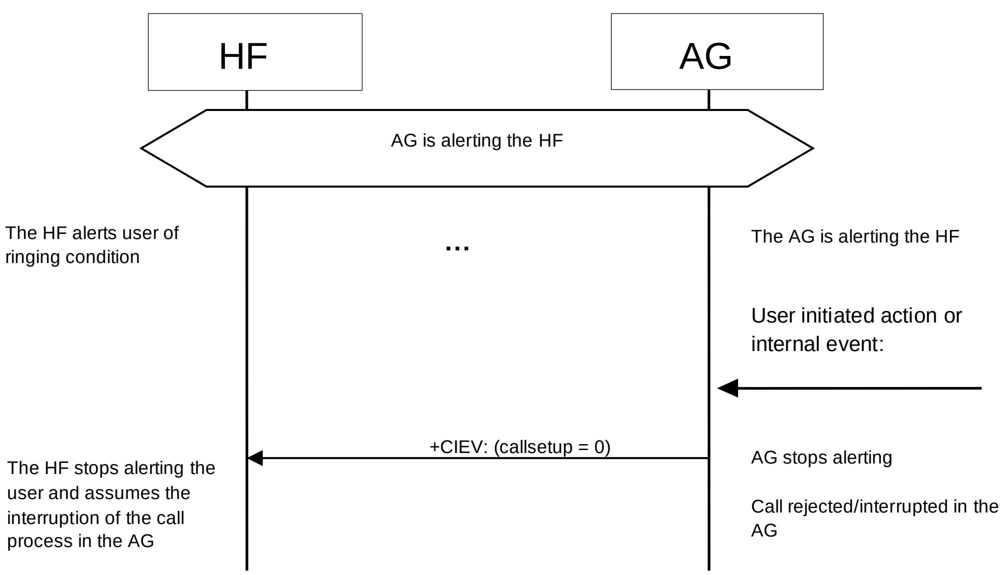

Figure 4.29: Rejection/interruption of an incoming call in the AG

4.15 Terminate a Call Process

An ongoing call process may be terminated by either the HF or the AG, by means of a user action or any other event.

4.15.1 Terminate a Call Process from the HF

The following preconditions apply for this procedure:

- An ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.
- A call-related process is ongoing in the AG.

Although not required for the call termination process, an Audio Connection is typically present between the HF and AG.

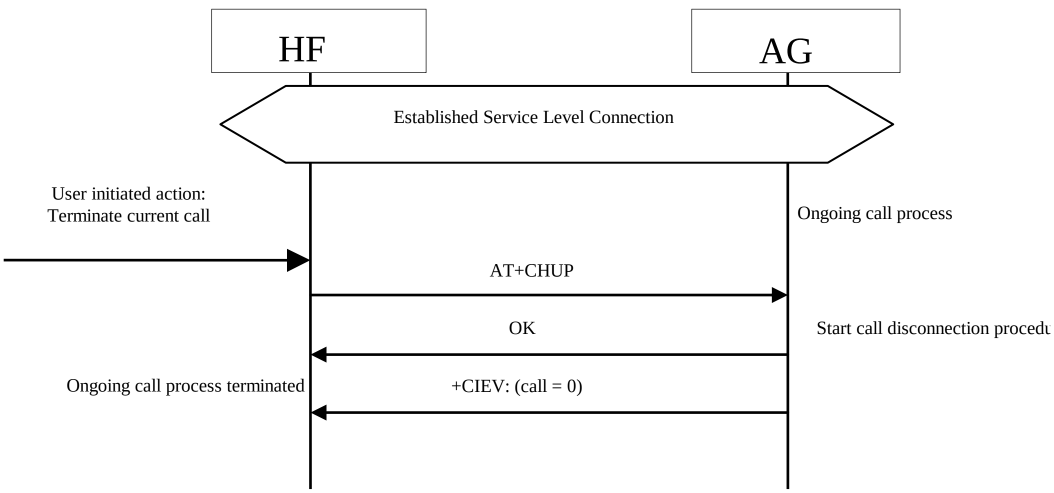

Figure 4.30: Terminate ongoing call - HF initiated

The user may abort the ongoing call process using whatever means provided by the Hands-Free unit. The HF shall send AT+CHUP command (see Section 5) to the AG, and the AG shall then start the procedure to terminate or interrupt the current call procedure. The AG shall then send the OK indication followed by the +CIEV result code, with the value indicating (call=0).

Performing a similar procedure, the AT+CHUP command described above may also be used for interrupting a normal outgoing call set-up process.

4.15.2 Terminate a Call Process from the AG

The following preconditions apply for this procedure:

- An ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the AG shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.
- A call related process is ongoing in the AG.

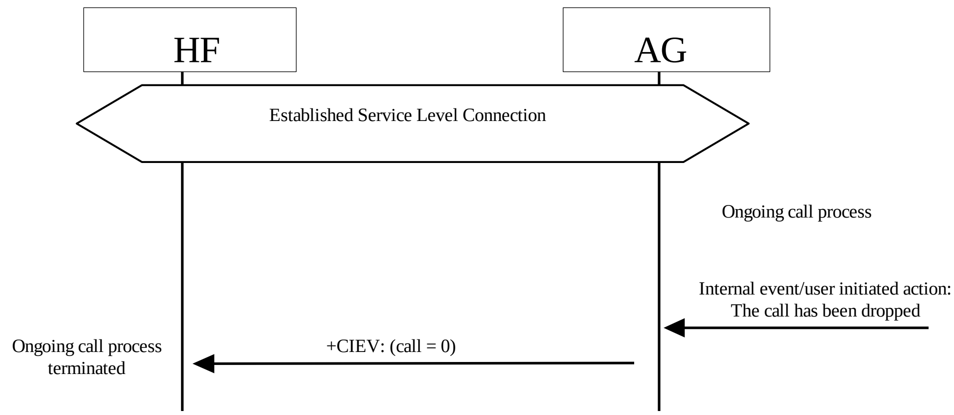

Figure 4.31: Terminate ongoing call - AG initiated

This procedure is fully applicable for cases in which an ongoing call process is interrupted in the AG for any reason.

In this case, the AG shall send the +CIEV result code, with the value indicating (call=0).

4.16 Audio Connection Transfer towards the HF

The audio paths of an ongoing call may be transferred from the AG to the HF. This procedure represents a particular case of an "Audio Connection set up" procedure, as described in Section 4.11.

The call connection transfer from the AG to the HF is initiated by user action either on the HF or on the AG. This shall result in either the HF or the AG, respectively, initiating an "Audio Connection set up" procedure with the audio paths of the current call being routed to the HF.

This procedure is only applicable if there is no current Audio Connection established between the HF and the AG. In fact, if the Audio Connection already exists, this procedure is not necessary because the audio path of the AG is assumed to be already routed towards the HF.

The following preconditions apply for this procedure:

- An ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the initiator of the "Audio Connection transfer towards the HF" procedure shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.
- An ongoing call exists in the AG, with the audio paths routed to the AG means.

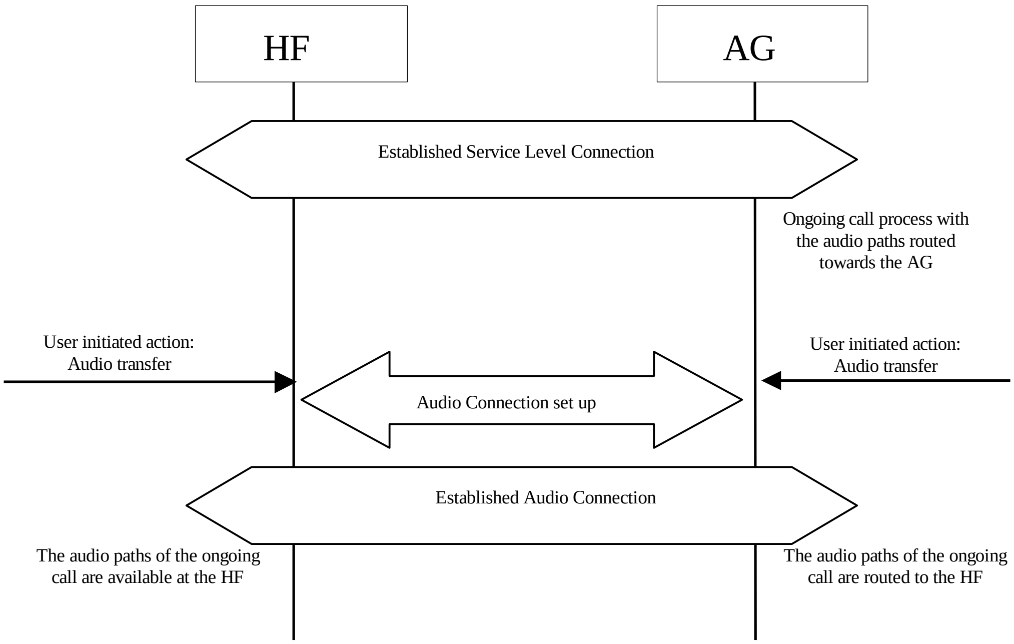

Figure 4.32: Audio Connection transfer to the HF

4.17 Audio Connection Transfer towards the AG

The audio paths of an ongoing call may be transferred from the HF to the AG. This procedure represents a particular case of an "Audio Connection release" procedure, as described in Section 4.12.

The call connection transfer from the HF to the AG is initiated by a user action in the HF or due to an internal event or user action on the AG side. This results in an "Audio Connection release" procedure being initiated either by the HF or the AG respectively, with the current call kept and its audio paths routed to the AG.

If, as a consequence of an HF initiated "Audio Connection transfer towards the AG" procedure, the existing Service Level Connection is autonomously removed by the AG, the AG shall attempt to re-establish the Service Level Connection once the current call ends.

As a precondition for this procedure, an ongoing call process shall exist in the AG. The audio paths of the ongoing call shall be available in the HF via an Audio Connection established between the AG and the HF.

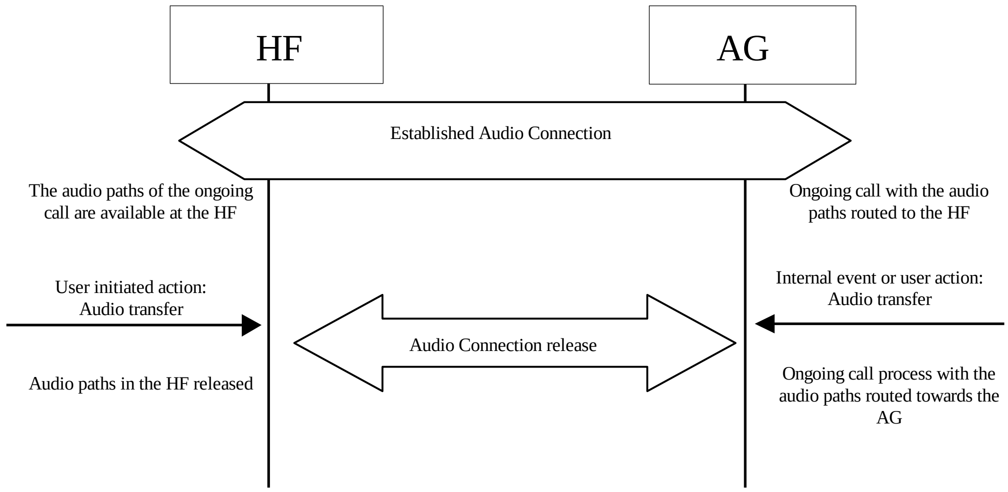

Figure 4.33: Audio Connection transfer to the AG

4.18 Place a Call with the Phone Number Supplied by the HF

The HF may initiate outgoing voice calls by providing the destination phone number to the AG. To start the call set-up, the HF shall initiate the Service Level Connection establishment (if necessary) and send a proper ATDdd…dd; command to the AG. The AG shall then start the call establishment procedure using the phone number received from the HF and issues the +CIEV result code, with the value (callsetup=2) to notify the HF that the call set-up has been successfully initiated.

See Section 5 for more information on the ATDdd…dd; command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

If an Audio Connection is not established, the AG shall establish the proper Audio Connection and route the audio paths of the outgoing call to the HF immediately following the commencement of the ongoing call set up procedure.

Once the AG is informed that the alerting of the remote party has begun, the AG shall issue the +CIEV result code, with the value indicating (callsetup=3). If the wireless network does not provide the AG of an indication of alerting the remote party, the AG may not send this indication.

Upon call connection the AG shall send the +CIEV result code, with the value indicating (call=1).

If the normal outgoing call establishment procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0), to notify the HF of this condition (see Section 4.15.2).

If the AG supports the “Three-way calling” feature and if a call is already ongoing in the AG, performing this procedure shall result in a new call being placed to a third party with the current ongoing call put on hold. For details on how to handle multiparty calls refer to Section 4.22.2.

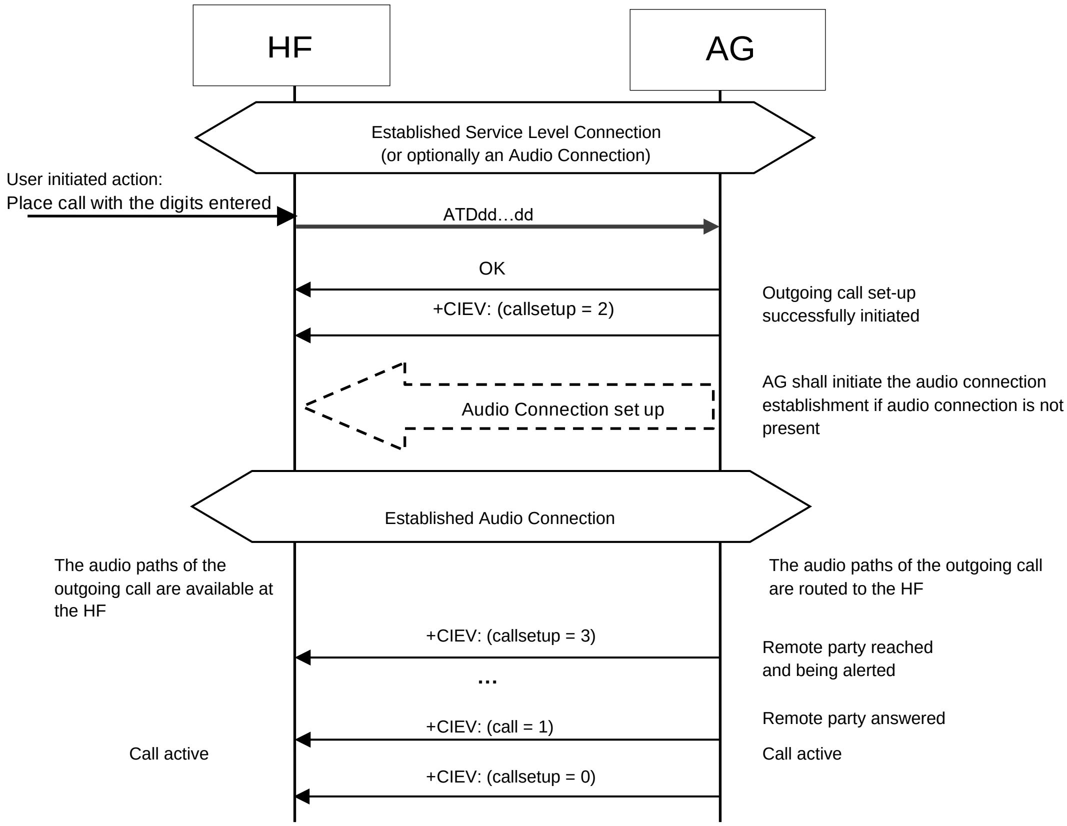

Figure 4.34: Place an outgoing voice call with the digits entered in the HF

4.19 Memory Dialing from the HF

The HF may initiate outgoing voice calls using the memory dialing feature of the AG. To start the call set-up, the HF shall initiate the Service Level Connection establishment (if necessary) and send an ATD>Nan…; command to the AG. The AG shall then start the call establishment procedure using the phone number stored in the AG memory location given by Nan…; and issue the +CIEV result code, with the value (callsetup=2) to notify the HF that the call set-up has been successfully initiated.

See Section 5 for more information on the ATD>Nan… command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

If an Audio Connection is not established, the AG shall establish the proper Audio Connection and route the audio paths of the outgoing call to the HF immediately following the commencement of the ongoing call set up procedure.

Once alerting of the remote party begins, the AG shall issue the +CIEV result code, with the value indicating (callsetup=3).

Upon call connection the AG shall send the +CIEV result code, with the value indicating (call=1).

If the normal outgoing call establishment procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0), to notify the HF of this condition (see Section 4.15.2).

If the AG supports the "Three-way calling" feature and if a call is already ongoing in the AG, performing this procedure shall result in a new call being placed to a third party with the current ongoing call put on hold. For details on how to handle multiparty calls refer to Section 4.22.2.

If there is no number stored for the memory location given by the HF, the AG shall respond with ERROR.

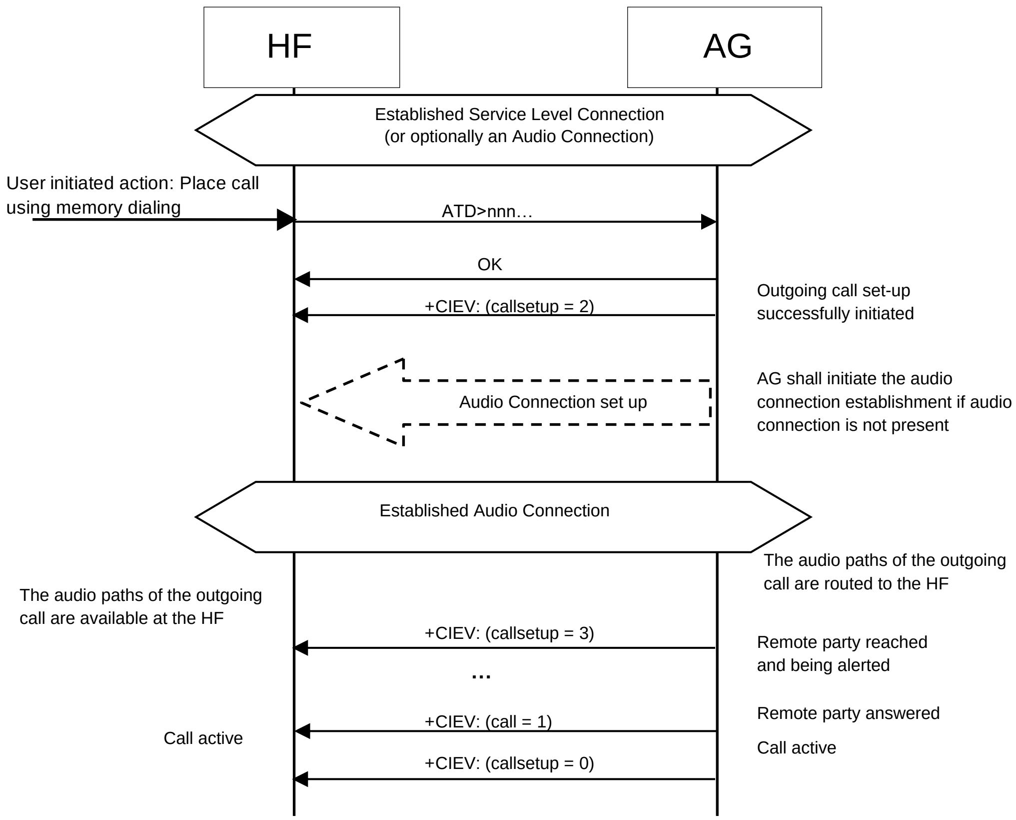

Figure 4.35: Place an outgoing voice call using memory dialing

4.20 Last Number Re-Dial from the HF

The HF may initiate outgoing voice calls by recalling the last number dialed by the AG. To start the call set-up, the HF shall initiate the Service Level Connection establishment (if necessary) and send an AT+BLDN command to the AG. The AG shall then start the call establishment procedure using the last phone number dialed by the AG, and issues the +CIEV result code, with the value (callsetup=2), to notify the HF that the call set-up has been successfully initiated.

See Section 5 for more information on the AT+BLDN command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

If an Audio Connection is not established, the AG shall establish the proper Audio Connection and route the audio paths of the outgoing call to the HF immediately following the commencement of the ongoing call set-up procedure.

Once alerting of the remote party begins, the AG shall issue the +CIEV result code, with the value indicating (callsetup=3).

Upon call connection the AG shall send the +CIEV result code, with the value indicating (call=1).

If the normal outgoing call establishment procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0), to notify the HF of this condition (see Section 4.15.2).

If the AG supports the "Three-way calling" feature and if a call is already ongoing in the AG, performing this procedure shall result in a new call being placed to a third party with the current ongoing call put on hold. For details on how to handle multiparty calls refer to Section 4.22.2.

If there is no number stored for the memory location given by the HF, the AG shall respond with ERROR.

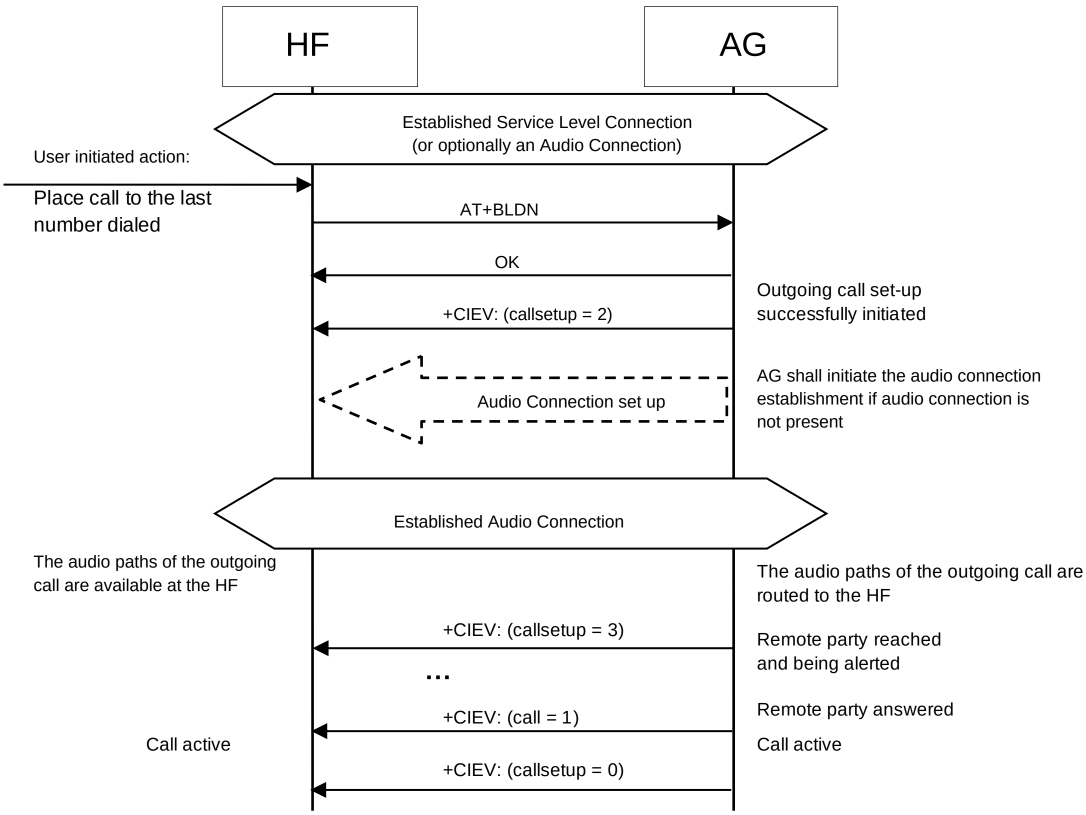

Figure 4.36: Place an outgoing voice call with the last number dialed

4.21 Call Waiting Notification Activation

The HF may issue the AT+CCWA command to enable the "Call Waiting notification" function in the AG. Once the "Call Waiting notification" is enabled, the AG shall send the corresponding +CCWA unsolicited result code to the HF whenever an incoming call is waiting during an ongoing call. It is always assumed that the "call waiting" service is already active in the network.

Once the HF issues the AT+CCWA command, the AG shall respond with OK. It shall then keep the "Call Waiting notification" enabled until either the AT+CCWA command is issued to disable "Call Waiting notification," or the current Service Level Connection between the AG and the HF is dropped for any reason.

See Section 5 for more information on the AT+CCWA command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

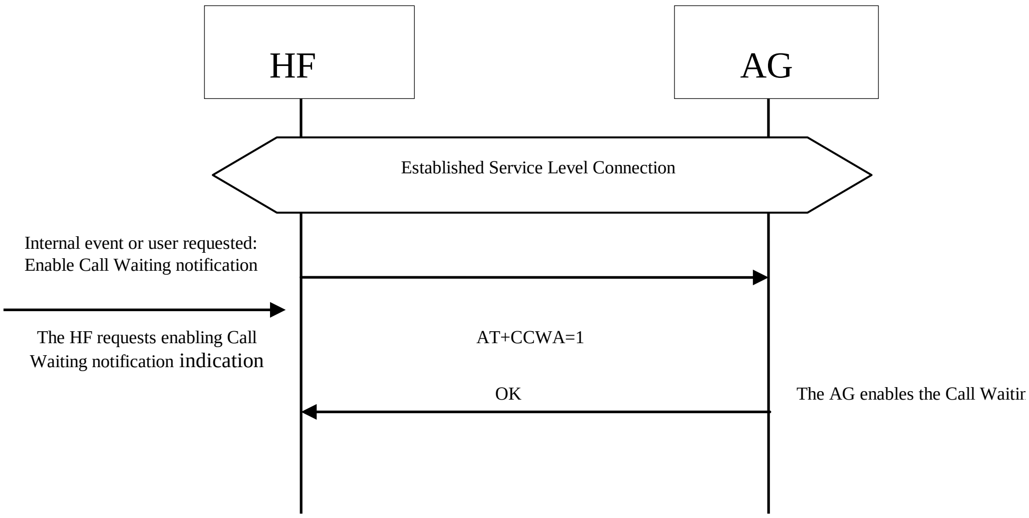

Figure 4.37: Activation of Call waiting notification

4.22 Three-Way Call Handling

Proper management of several concurrent calls shall be accomplished by performing the procedures described in [2] but with some limitations stated in this specification. For more details, refer to Section 5.

The HF device cannot always assume that the "call hold and/or multiparty" services are available in the network. If the AG determines that a requested action by the HF device cannot be performed due to the inability of the network to support that feature or lack of subscriber subscription, the AG shall return a +CME error.

There are two +CME ERROR codes that are used to indicate network related failure reasons to the HF:

- 30 - No Network Service. Indicates that an AT+CHLD command cannot be implemented due to network limitations.
- 31 - Network Timeout. Indicates that an AT+CHLD command cannot be implemented due to network problems.

In general, when the user deals with multiple concurrent calls, the HF shall issue the corresponding AT+CHLD command as a result of user actions. This command allows the control of multiple concurrent calls and provides means for holding calls, releasing calls, switching between two calls, and adding a call to a multiparty conference.

When this feature is supported, the HF and AG are only mandated to implement the “basic Three-Way calling” commands AT+CHLD = 1 and 2.

This section covers two cases. In one case, the third party call is received in the AG, and notification is sent to the HF via a Call Waiting notification. In the second case, the third party call is placed from the HF.

See Section 5 for more information on the AT+CHLD command.

The following preconditions apply for these procedures:

- As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the initiator of the procedure shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.
- An ongoing call in the AG shall exist.

4.22.1 Three-Way Calling—Call Waiting Notification

In addition to the two previously stated preconditions, the Call Waiting notification to the HF shall already be enabled in the AG (that is, the procedure stated in Section 4.21 has been performed).

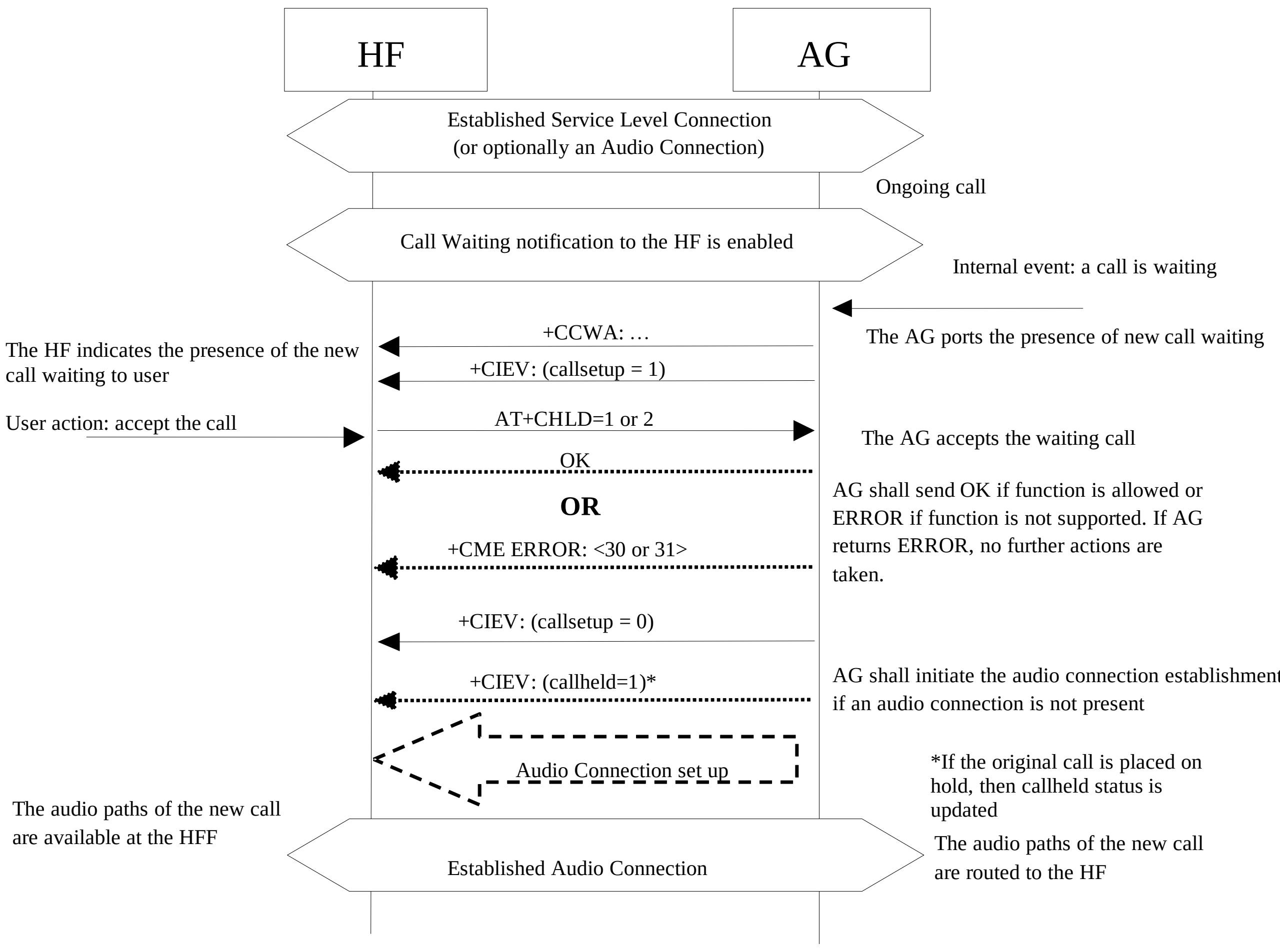

Figure 4.38: Typical Call Waiting indication followed by HF possible response

If the AG receives a third party call, it shall send the call waiting notification +CCWA and +CIEV result code, with the value indicating (callsetup=1), to the HF.

If the user rejects the call at the HF, the HF shall send the AT+CHLD command with parameter 0 to the AG. The AG shall then reject the call and respond with OK, and issue the +CIEV result code with the value indicating (callsetup=0).

If the user accepts the call at the HF, the HF shall send the AT+CHLD with parameter 1 or 2 to the AG. The HF cannot cause the waiting call to be added as a conference call via a single AT+CHLD command; but if this is desired the HF can achieve this by first issuing an AT+CHLD=2 command, and then issuing an AT+CHLD=3 command. The AG shall then accept the waiting call and respond with OK, and issue the +CIEV result code with the value indicating (callsetup=0). If the HF elects to send AT+CHLD=2 (placing the original call on hold), then the AG shall send the +CIEV result code with the value indicating a held call (callheld=1).

Optionally, the HF may then use the AT+CHLD command, in order to change the status of the held and active calls.

If the normal incoming call procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0), to notify the HF of this condition (see Section 4.14.2).

4.22.2 Three-Way Calls – Third Party Call Placed from the HF

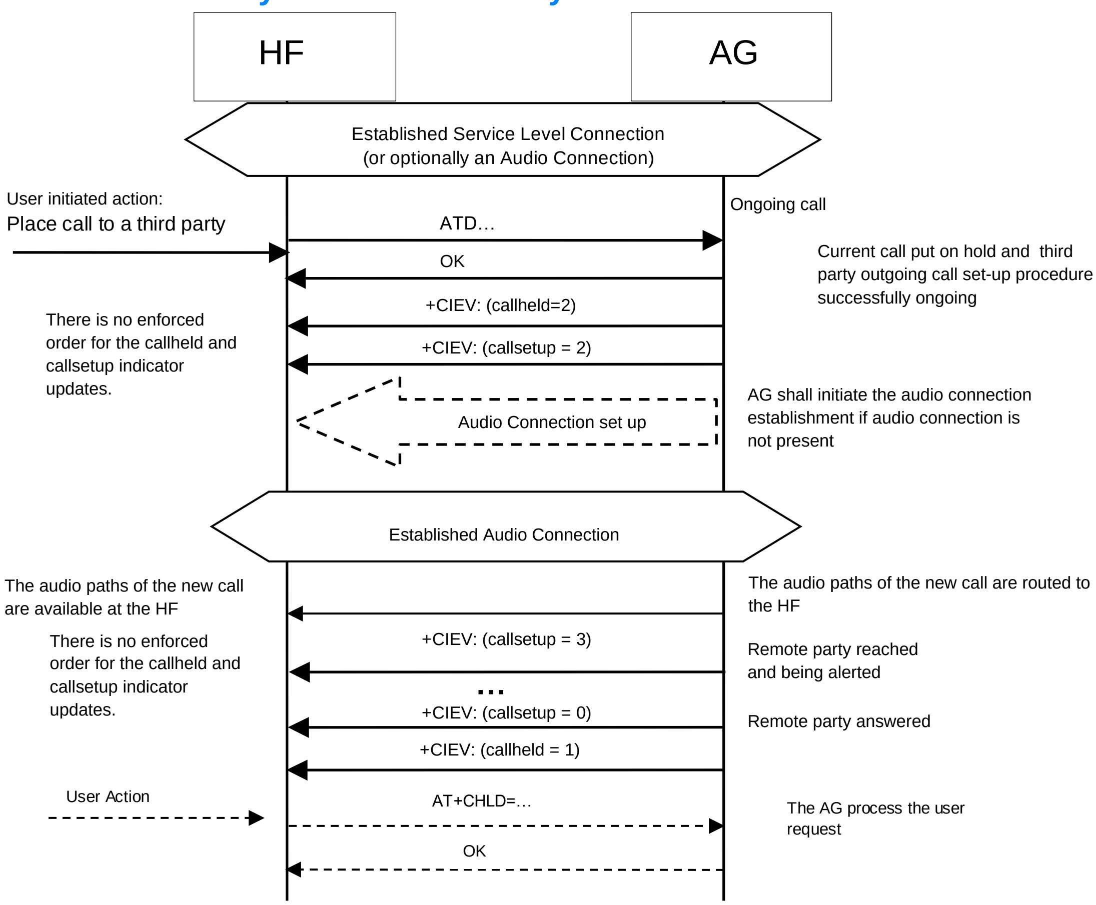

Figure 4.39: Three-way call handling when the third party call is placed from the HF

If a third party call is placed from the HF using the ATD command, the AG shall send the OK indication and two +CIEV result codes, one with the value indicating (callsetup=2), and one with the value indicating (callheld=2) to the HF. It is permissible for the AG to send these two +CIEV result codes in either order as the timing of events in the AG may differ between implementations and network types. If the remote party is reached and alerted, the AG shall issue the +CIEV result code with the value indicating (callsetup=3). If the wireless network does not provide the AG of an indication of alerting the remote party, the AG may not send this indication.

If the remote party answers the call, the AG shall issue the +CIEV result codes with the values indicating (callsetup=0) and (callheld=1). As before, there is no enforced order to these two +CIEV result codes.

Optionally, the HF may then use the AT+CHLD command in order to change the status of the held and active calls. If the AT+CHLD command results in the change in a held call status the AG shall provide the status indication using the +CIEV result code with the value indicating the call held status (callheld=<0,1,2>).

If the normal outgoing call procedure is interrupted for any reason, the AG shall issue the +CIEV result code, with the value indicating (callsetup=0), to notify the HF of this condition (see Section 4.15.2). The AG shall then update the callheld status to indicate the change of status of the original (held) call based upon one of the two following scenarios:

- The AG may choose to leave the original call on hold. In this case, the AG shall issue the +CIEV result code with the value indicating (call held=2).
- Alternatively the AG may autonomously retrieve the held call, thus changing the status and shall send the +CIEV indicator (call held=0).

In either case, the +CIEV response code (call=1) shall remain unchanged.

4.23 Calling Line Identification (CLI) Notification

The HF may issue the AT+CLIP command to enable the "Calling Line Identification notification" function in the AG.

If the calling subscriber number information is available from the network, the AG shall issue the +CLIP unsolicited result code just after every RING indication when the HF is alerted in an incoming call. See Section 4.13 for more details.

Once the HF issues the AT+CLIP command, the AG shall respond with OK. The AG shall then keep the "Calling Line Identification notification" enabled until either the AT+CLIP command is issued by the HF to disable it, or the current Service Level Connection between the AG and the HF is dropped for any reason.

See Section 5 for more information on the AT+CLIP command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

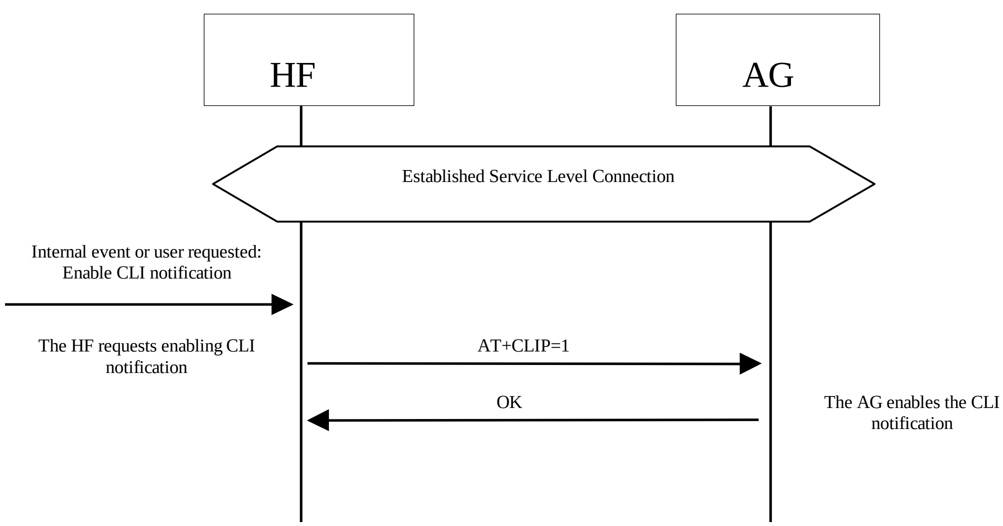

Figure 4.40: Activation of CLI notification

4.24 The HF Requests Turning off the AG's EC and NR

The HF may disable the echo canceling and noise reduction functions resident in the AG via the AT+NREC command.

If the HF supports embedded EC and/or NR functions, it shall support the AT+NREC command as described in the procedures in this section. Moreover, if the HF has these functions enabled, it shall perform this procedure before any Audio Connection between the HF and the AG is established.

By default, if the AG supports its own embedded echo canceling and/or noise reduction functions, it shall have them activated until the AT+NREC command is received. From then on, and until the current Service Level Connection between the AG and HF is dropped for any reason, the AG shall disable these functions every time an Audio Connection between the HF and the AG is used for audio routing.

If the AG does not support any echo canceling and noise reduction functions, it shall respond with the ERROR indicator on reception of the AT+NREC command.

See Section 5 for more information on the AT+NREC command.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

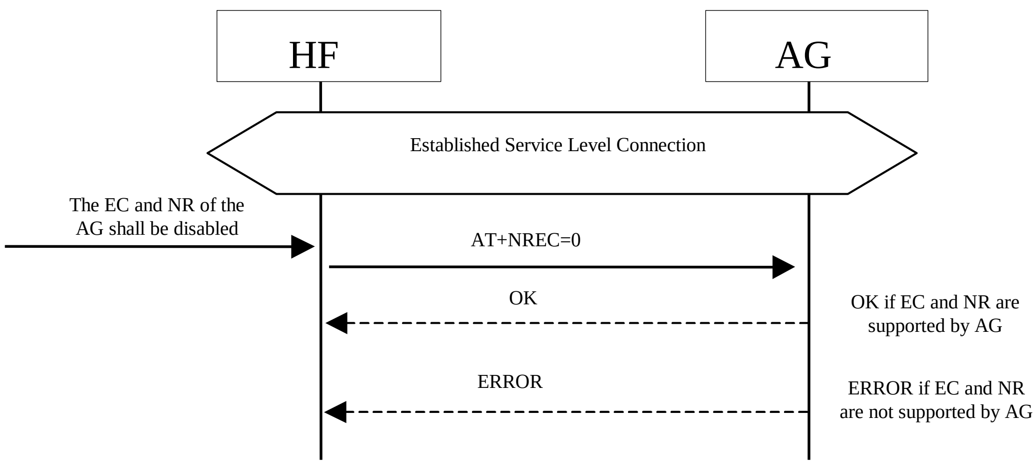

Figure 4.41: NR and EC functions available in the AG

The HF sends the AT+NREC command and AG confirms with either OK or ERROR indication.

4.25 Voice Recognition Activation / Enhanced Voice Recognition Activation

The HF, via the AT+BVRA command, or the AG autonomously, may activate/deactivate the voice recognition function resident in the AG. The Enhanced Voice Recognition Status and Voice Recognition Text features enhance the use of the AT+BVRA command. This is described in detail in Section 5.3. Beyond the audio routing and voice recognition activation capabilities, the rest of the voice recognition functionality is implementation dependent.

Whenever the AG supports a voice recognition function, it shall support the AT+BVRA command as described in the procedures in this section.

If the HF issues the AT+BVRA command, the AG shall respond with the OK result code if it supports voice recognition, then initiate an Audio Connection to the HF (if the Audio Connection does not already exist) and begin the voice input sequence.

If the AG does not support voice recognition, the AG shall respond with the ERROR indication.

When the voice recognition function is activated from the AG, it shall inform the HF via the +BVRA: 1 unsolicited result code and the AG shall initiate an Audio Connection to the HF (if the Audio Connection does not already exist) and begin the voice input sequence.

Once activated, depending upon the voice recognition implementation, the AG shall then keep the voice recognition function enabled:

- For the duration of time supported by the implementation (“momentary on” voice recognition implementation). In this case, the AG shall notify the HF by sending a +BVRA: 0 unsolicited result code.
- Or until the AT+BVRA command is issued to disable voice recognition from the HF.
- Or until the current Service Level Connection between the AG and the HF is dropped for any reason.

See Section 5 for more information on the AT+BVRA command and the +BVRA result code.

If the Enhanced Voice Recognition Status or Voice Recognition Text features are supported, the AT+BVRA command is extended with the value 2. The value 2 shall only be used if both the AG and the HF support the Enhanced Voice Recognition Status feature. It indicates that the HF is ready to accept audio when the Audio Connection is first established.

The HF may send this value during an ongoing VR (Voice Recognition) session to terminate audio output from the AG (if there is any) and prepare the AG for new audio input.

The syntax of the +BVRA command is extended with <vrecstate> and <textualRepresentation>.

For example, <vrecstate> of 5 (b101) means that the AG is listening to the audio input and processing at the same time.

Examples that build on one another for textual representation:

1. +BVRA: 1,1,12AA,1,1,“Message 1 from Janina”: A new text is sent from the AG to the HF. The AG now has the text:
   12AA: Message 1 from Janina

2. +BVRA: 1,1,12AB,1,1,“Message 2 from Melissa”: A new text is sent from the AG to the HF with a new <textID>. The AG now has the text:
   12AA: Message 1 from Janina
   12AB: Message 2 from Melissa

3. +BVRA: 1,1,12AB,1,2 ,“Message 3 from Melissa”: A new text is sent from the AG to the HF with the same <textID> as example 2. Since the textID is the same as in example 2 and the operation is replace, the text from example 2 is replaced by the new text. So the AG now has the texts:
   12AA: Message 1 from Janina
   12AB: Message 3 from Melissa

4. +BVRA: 1,1,12AB,1,3,“ with new stuff”: A new text is sent from the AG to the HF and gets attached to the old text. Since the textID is the same as in example 3 and the operation is append, the new text is appended to the text from example 3. The AG now has the texts:
   12AA: Message 1 from Janina
   12AB: Message 3 from Melissa with new stuff

For more information, refer to Section 5.3.

A precondition for these procedures is that an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the initiator of the procedure shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

4.25.1 Voice Recognition Activation – HF Initiated

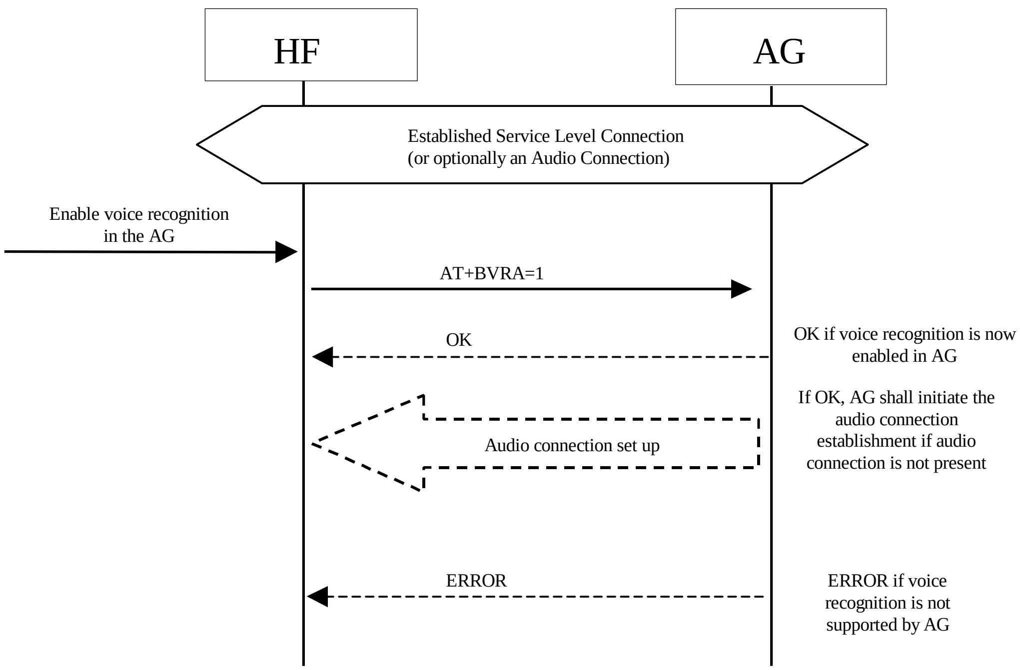

Figure 4.42: Voice recognition activation – HF initiated

4.25.2 Voice Recognition Activation – AG Initiated

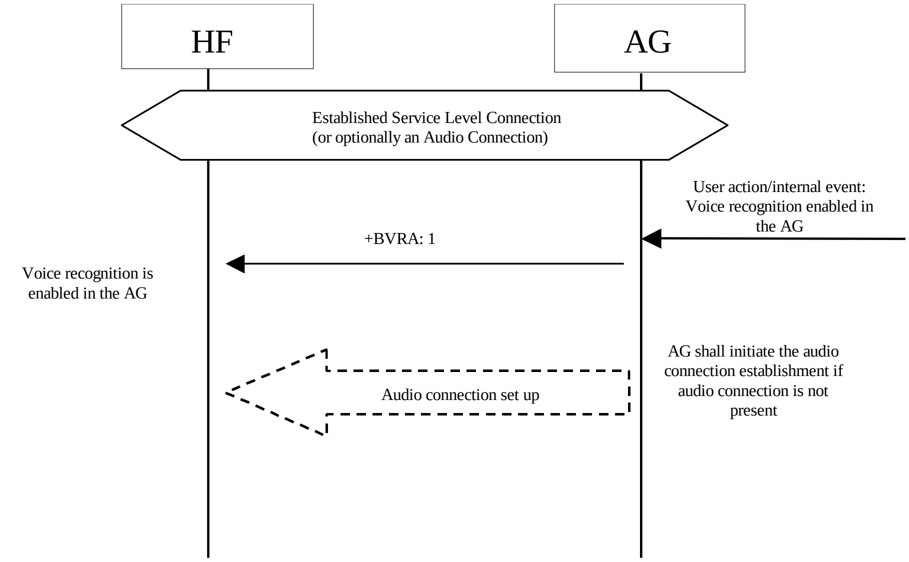

Figure 4.43: Voice recognition activation – AG initiated

4.25.3 Voice Recognition Deactivation

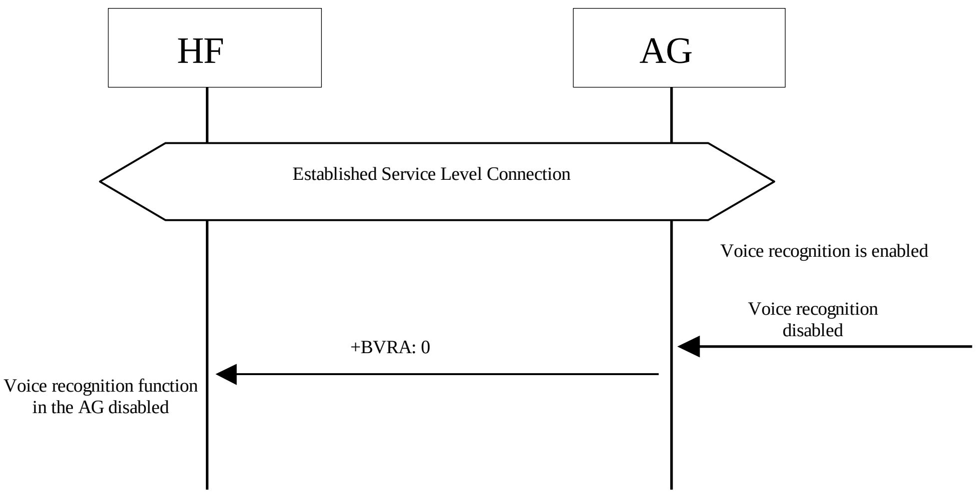

Figure 4.44: Voice recognition deactivation – "momentary on" approach

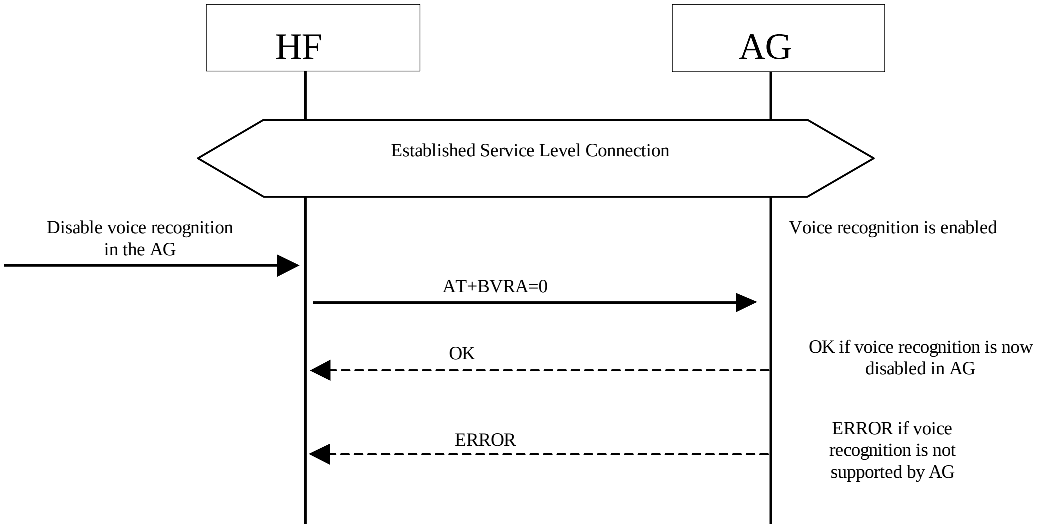

Figure 4.45: Voice recognition deactivation from the HF

4.25.4 Enhanced Voice Recognition Activation session

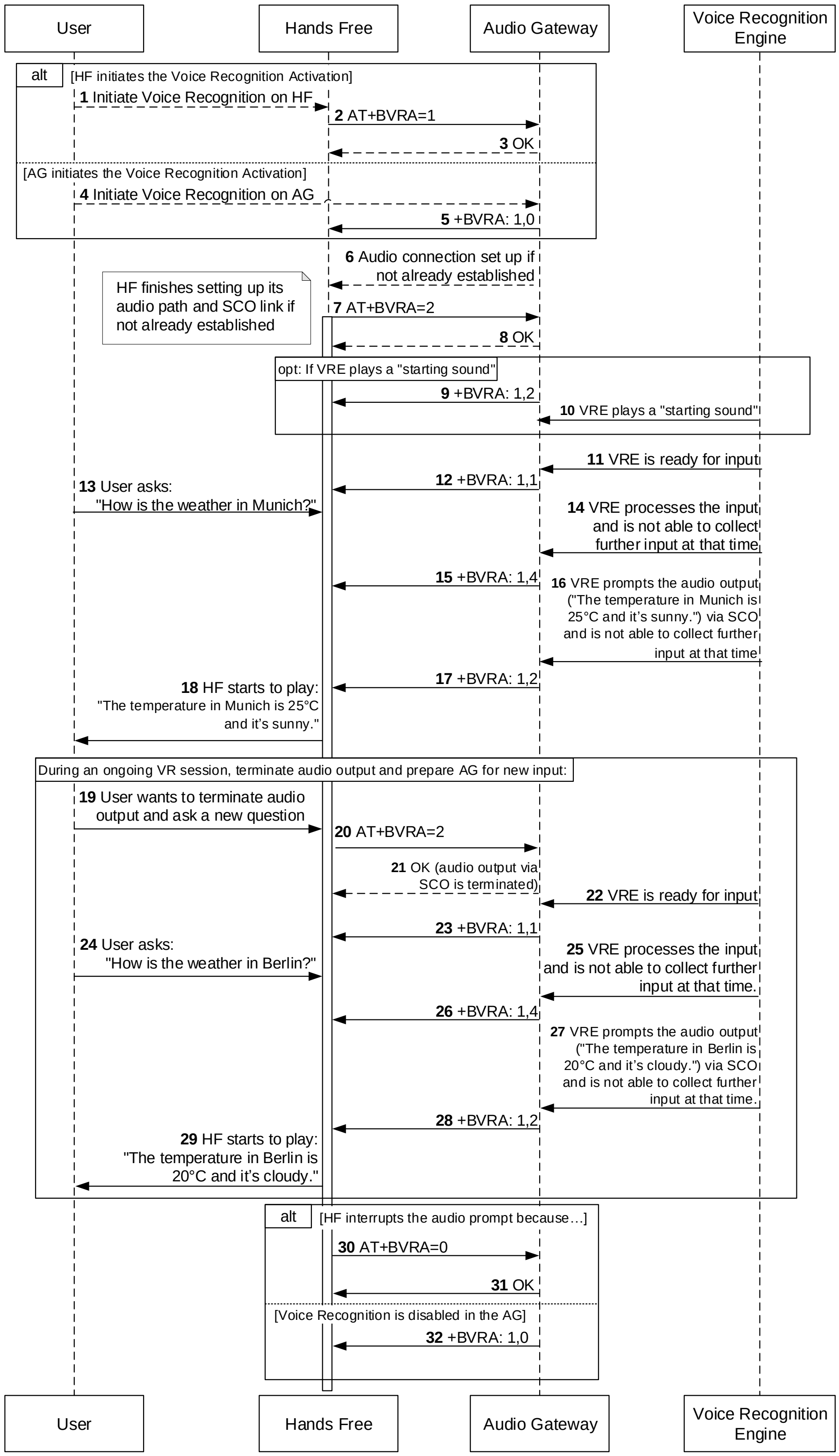

Figure 4.46: Example sequence diagram of an Enhanced Voice Recognition Activation session

4.26 Enhanced Voice Recognition Activation with textual representation

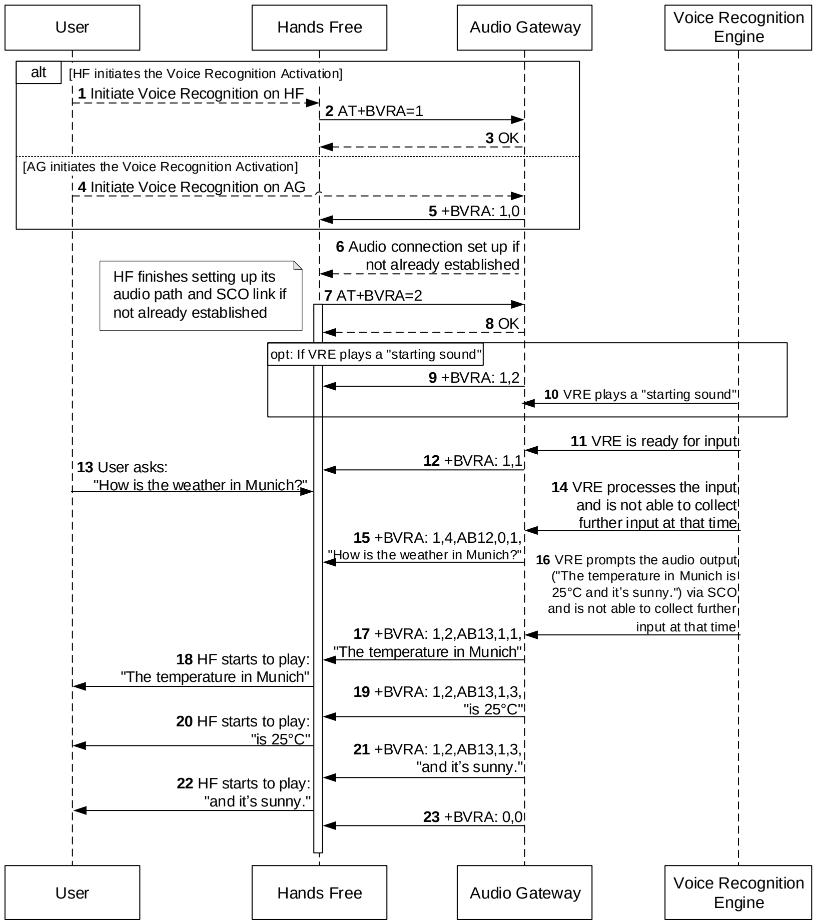

Figure 4.47: Sequence diagram with textual representation

4.27 Attach a Phone Number for a Voice Tag

This procedure is applicable to HFs supporting internal voice recognition functionality. It provides a means to read numbers from the AG for the purpose of creating a unique voice tag and storing the number and its linked voice tag in the HF's memory. The HF may then use its internal Voice Recognition to dial the linked phone numbers when a voice tag is recognized by using the procedure "Place a call with the phone number supplied by the HF" described in Section 4.18.

Upon an internal event or user action, the HF may request a phone number from the AG by issuing the AT+BINP=1 command. Depending on the current status of the AG, it may either accept or reject this request.

If the AG accepts the request, it shall obtain a phone number and send the phone number back to the HF by issuing the +BINP response.

If the AG rejects the request from the HF, it shall issue the ERROR result code to indicate this circumstance to the HF.

When this procedure is executed multiple times (to retrieve multiple AG phone numbers to be linked to voice tags), it is the responsibility of the AG to provide the next phone number to be passed to the HF each time the procedure is executed.

See Section 5 for more information on the AT+BINP command and the +BINP response.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

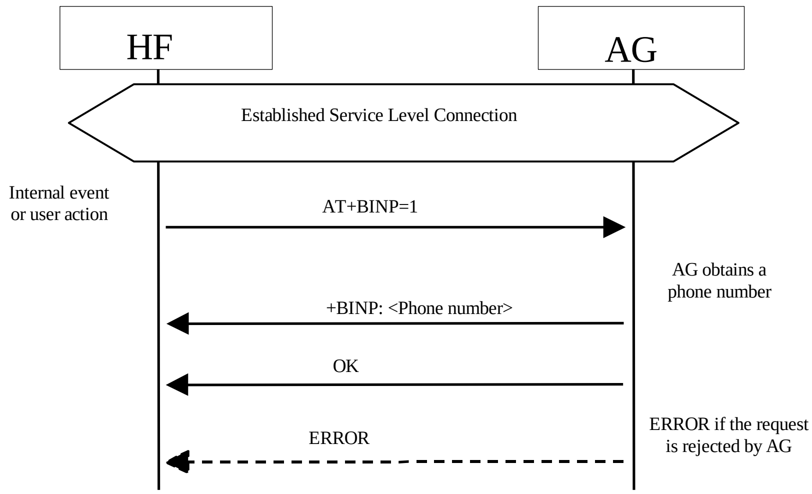

Figure 4.48: Request phone number to the AG

4.28 Transmit DTMF Codes

During an ongoing call, the HF transmits the AT+VTS command to instruct the AG to transmit a specific DTMF code to its network connection.

See Section 5 for more information on the AT+VTS command.

The following preconditions apply for this procedure:

- An ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.
- An ongoing call in the AG exists.

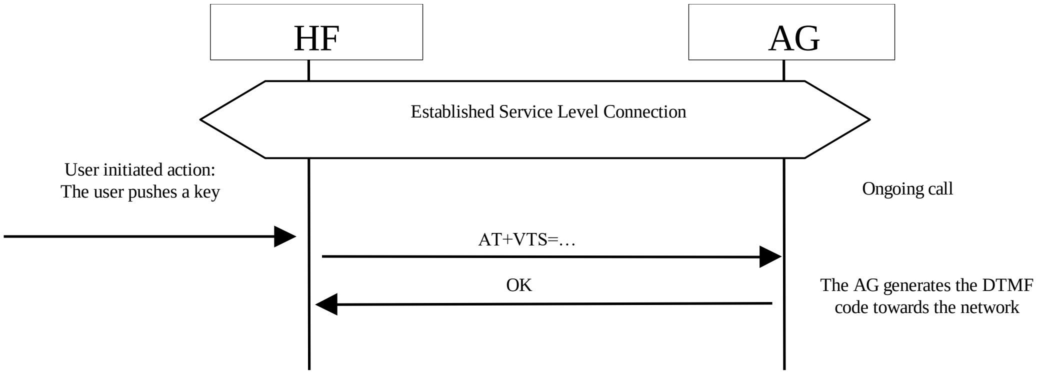

Figure 4.49: Transmit DTMF code

4.29 Remote Audio Volume Control

4.29.1 Audio Volume Control

This procedure enables the user to modify the speaker volume and microphone gain of the HF from the AG.

The AG may control the gain of the microphone and speaker of the HF by sending the unsolicited result codes +VGM and +VGS respectively. There is no limit in the amount and order of result codes.

If the remote audio volume control feature is supported in the HF device, it shall support at least remote control of the speaker volume.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the AG shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

An audio connection is not a necessary precondition for this feature.

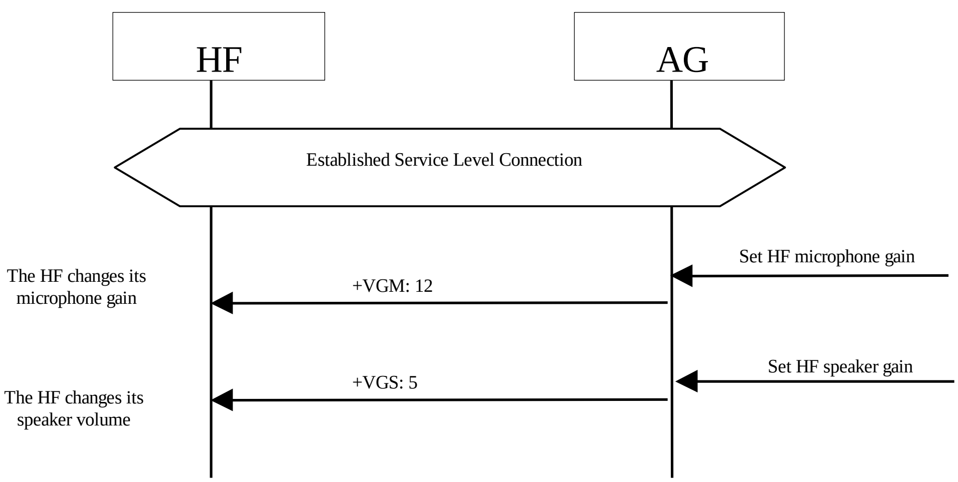

Figure 4.50: Typical example of audio volume control

Both the speaker and microphone gains are represented as parameter to the +VGS and +VGM, on a scale from 0 to 15. The values are absolute values, and relate to a particular (implementation dependent) volume level controlled by the HF.

See Section 5 for more information on these commands and unsolicited result codes.

4.29.2 Volume Level Synchronization

This procedure allows the HF to inform the AG of the current gain settings corresponding to the HF's speaker volume and microphone gain.

On Service Level Connection establishment, the HF shall always inform the AG of its current gain settings by using the AT commands AT+VGS and AT+VGM.

If local means are implemented on the HF to control the gain settings, the HF shall also use the AT commands AT+VGS and AT+VGM to permanently update the AG of changes in these gain settings.

In all cases, the gain settings shall be kept stored, at both sides, for the duration of the current Service Level Connection. Moreover, if the Service Level Connection is released as a consequence of an HF initiated "Audio Connection transfer towards the AG" as stated in Section 4.17, the HF shall also keep the gain settings and re-store them when the Service Level Connection is re-established.

The HF shall support speaker gain synchronization when it supports remote speaker gain control.

The HF shall support microphone gain synchronization when it supports remote microphone gain control.

As a precondition for this procedure, an ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

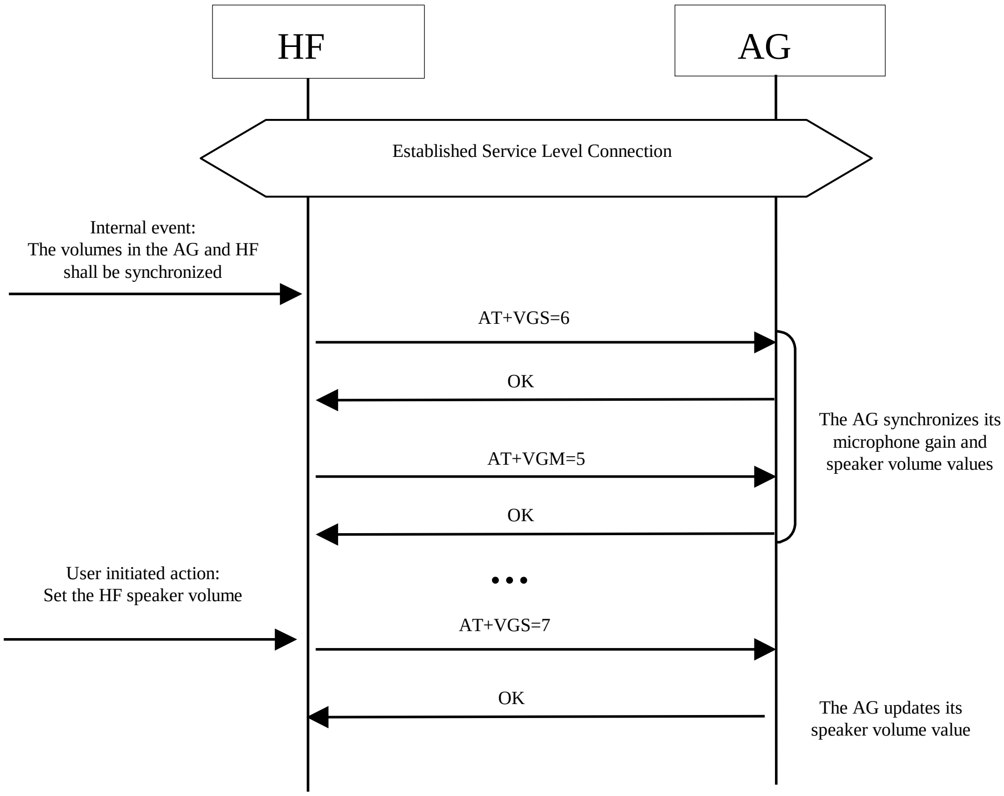

Figure 4.51: Typical example of volume level synchronization

See Section 5 for more information on these commands and unsolicited result codes.

4.30 Response and Hold

This procedure allows the user to put an incoming call on hold and then accept or reject the call from the HF or AG. This feature is specific to the limited markets where PDC and CDMA networks support this function.

4.30.1 Query Response and Hold Status

The HF shall execute this procedure to query the status of the "Response and Hold" state of the AG.

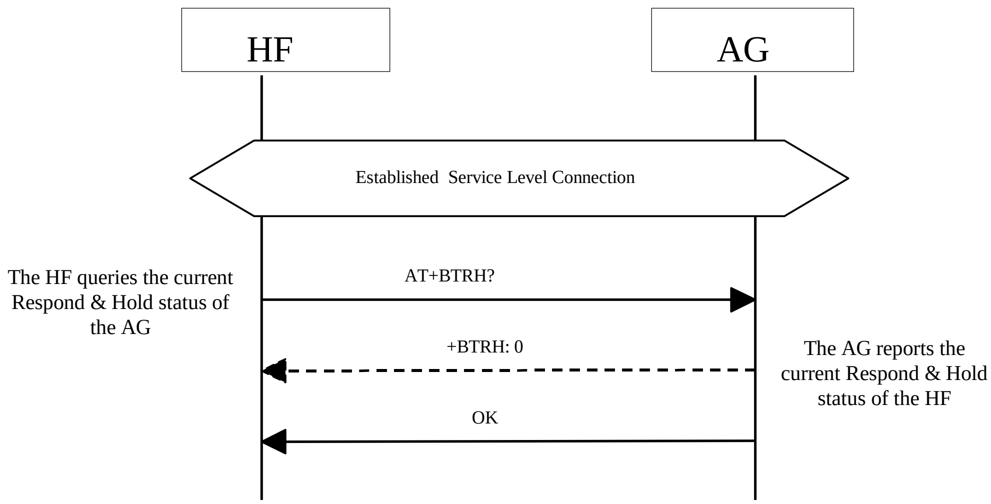

Figure 4.52: Query Response and Hold State of AG

- The HF shall issue AT+BTRH? command to query the current "Response and Hold" state of the AG.
- If the AG is currently in any of the Response and Hold states, then the AG shall send a +BTRH: Response with the parameter set to 0. If the AG is not in the Response and Hold states, then no response shall be sent.
- The AG shall send OK response to signal completion of the AT+BTRH? command.

4.30.2 Put an Incoming Call on Hold from HF

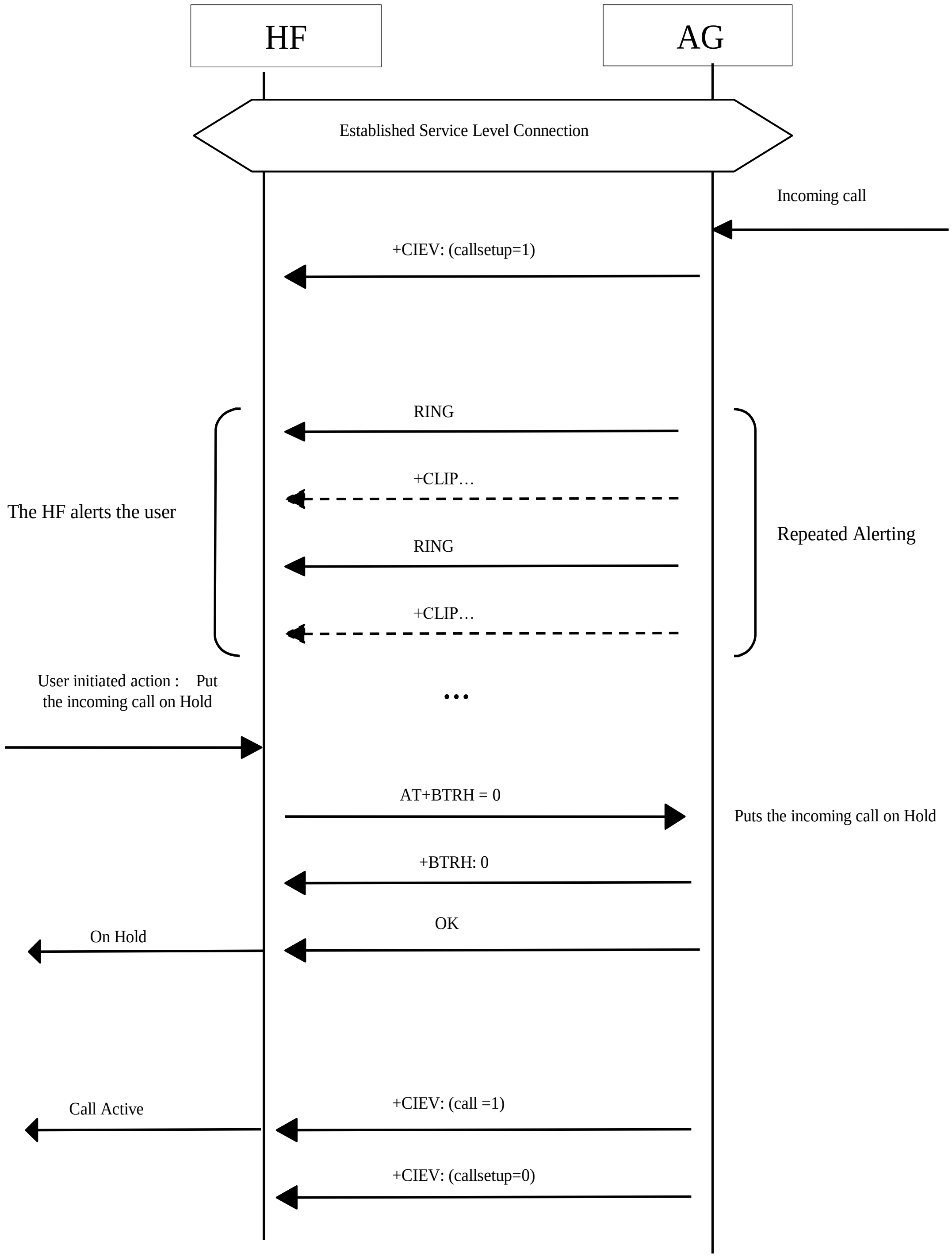

Figure 4.53: Put an incoming call on Hold from HF

- As a precondition to this procedure, the AG shall not have an active call or a call on hold.
- The AG shall send a sequence of unsolicited RING alerts to the HF. The RING alert shall be repeated until the HF accepts the incoming call or until the incoming call is interrupted for any reason.

- If the HF has enabled the Calling Line Identification [CLI], the AG shall send a +CLIP Response to HF.
- The user may put the incoming voice call on hold by using the proper means provided by the HF. The HF shall then send the AT+BTRH command with the parameter <n> set to 0. The AG shall then begin the procedure for putting the incoming call on hold.
- The AG shall send +BTRH Response with the parameter set to 0 as soon as the incoming call is put on hold.
- The +CIEV: (callheld = 2) message shall not be sent when a call is held via the AT+BTRH=0 message.
- The AG shall send the +CIEV Response with the call status set to 1.
- The AG shall send the +CIEV Response with the callsetup status set to 0.

4.30.3 Put an Incoming Call on Hold from AG

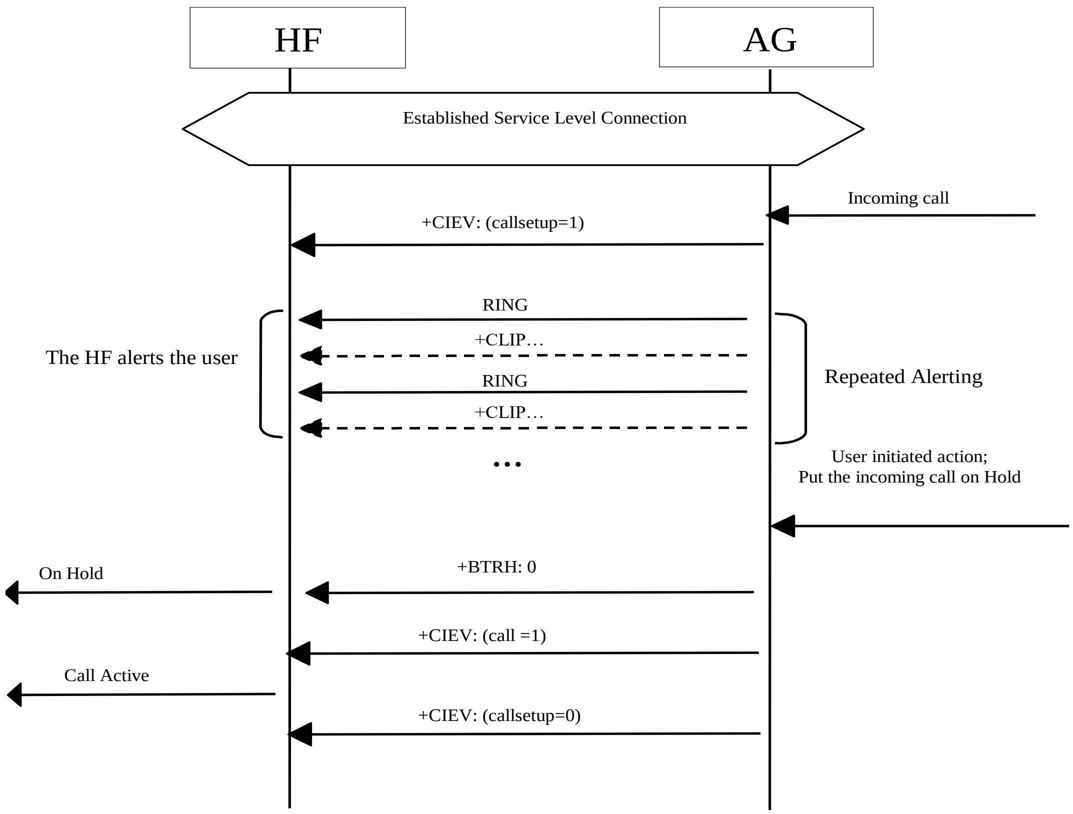

Figure 4.54: Put an incoming call on Hold from AG

As a precondition to this procedure, the AG shall not have an active call or a call on hold.

- The AG shall send a sequence of unsolicited RING alerts to the HF. The RING alert shall be repeated until the HF accepts the incoming call or until the incoming call is interrupted for any reason.

- If the HF has enabled the Calling Line Identification [CLI], the AG shall send a +CLIP Response to the HF.
- The user may put the incoming voice call on hold by using the proper means provided by the AG unit. The AG shall then send +BTRH Response with the parameter <n> set to 0 to indicate that the incoming call is on hold.
- The +CIEV: (callheld = 2) message shall NOT be sent by the audio gateway when it holds a call via the response and hold method.
- Depending on whether in-band ringing is enabled or disabled, there may or may not be a synchronous connection established between the HF and AG. The synchronous connection state (enabled or disabled) shall not be changed when an incoming call is placed on hold.
- The AG shall send the +CIEV Response with the call status set to 1.
- The AG shall send the +CIEV Response with the callsetup status set to 0.

4.30.4 Accept a Held Incoming Call from HF

The following additional precondition applies to this procedure:

- An incoming call was put on hold.

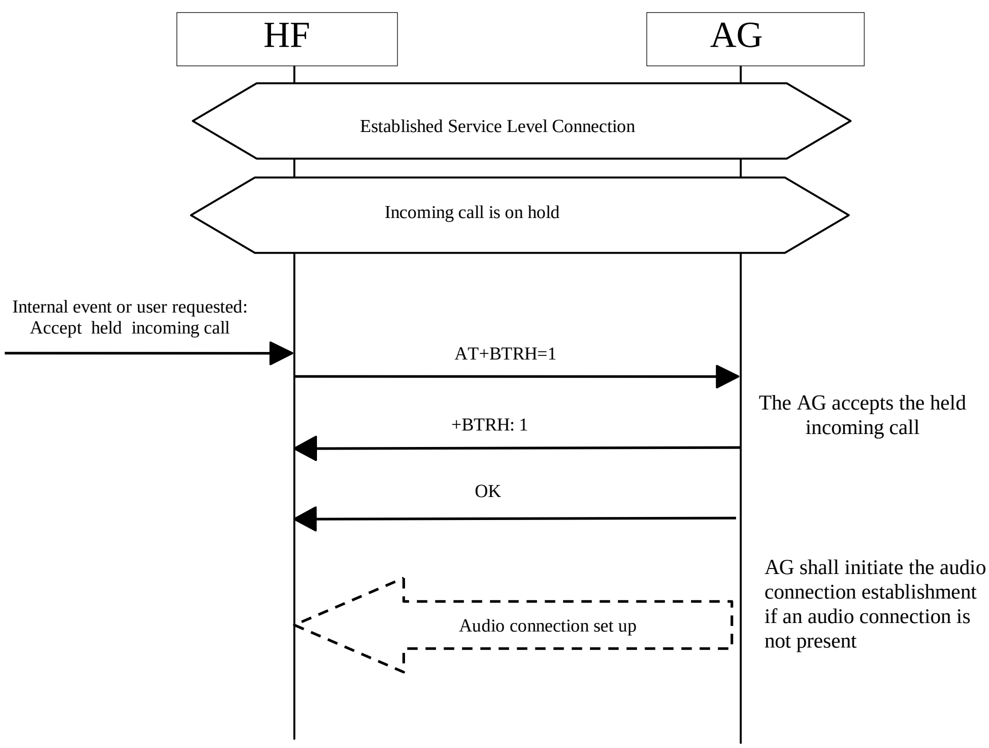

Figure 4.55: Accept a held incoming call from HF

- The user may accept the incoming voice call on hold by using the proper means provided by the HF. The HF shall then send the AT+BTRH command with the parameter <n> set to 1. The AG shall then begin the procedure for accepting the incoming call that was put on hold.

- The AG shall then send +BTRH Response with the parameter <n> set to 1 to notify HF that the held incoming call was accepted.
- The AG shall start the procedure for establishing the audio connection and route the audio paths to the HF only if the audio connection was not established.

4.30.5 Accept a Held Incoming Call from AG

The following additional precondition applies to this procedure:

- An incoming call was put on hold.

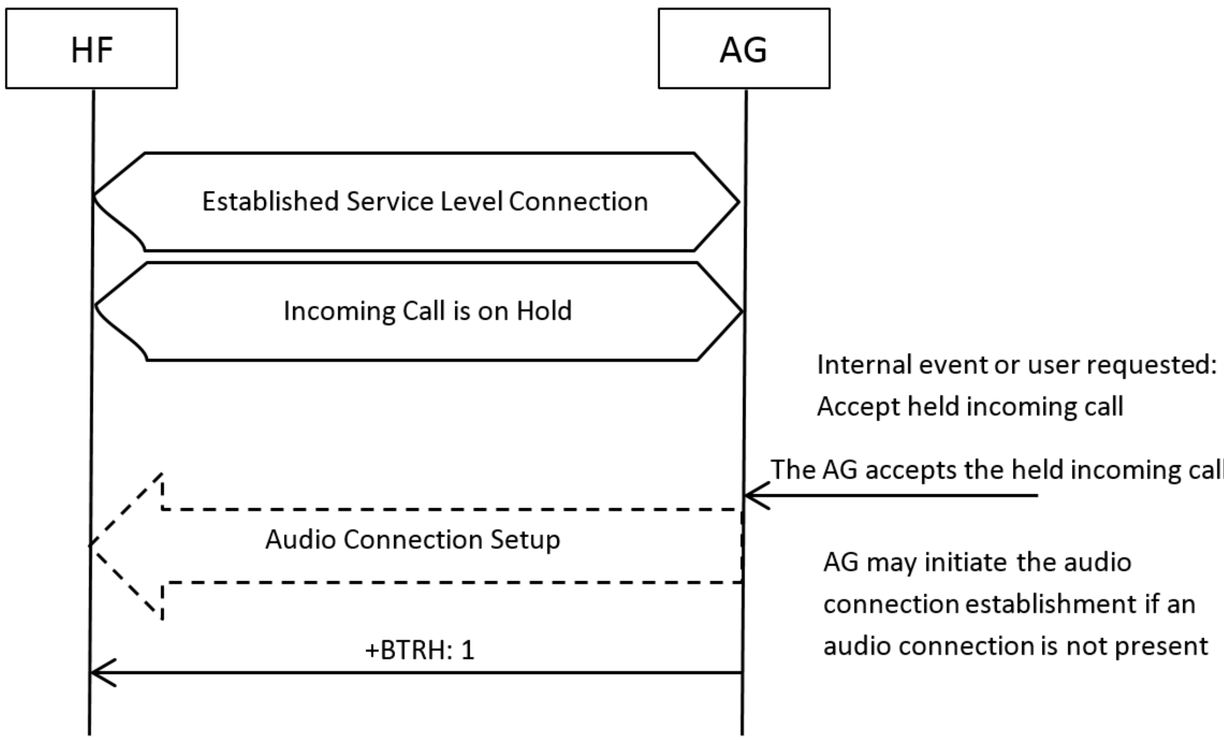

Figure 4.56: Accept a held incoming call from AG

- The user may accept the incoming voice call on hold by using the proper means provided by the AG unit. The AG shall then send +BTRH Response with the parameter <n> set to 1 to notify the HF that the held incoming call was accepted.

4.30.6 Reject a Held Incoming Call from HF

The following additional precondition applies to this procedure:

- An incoming call was put on hold.

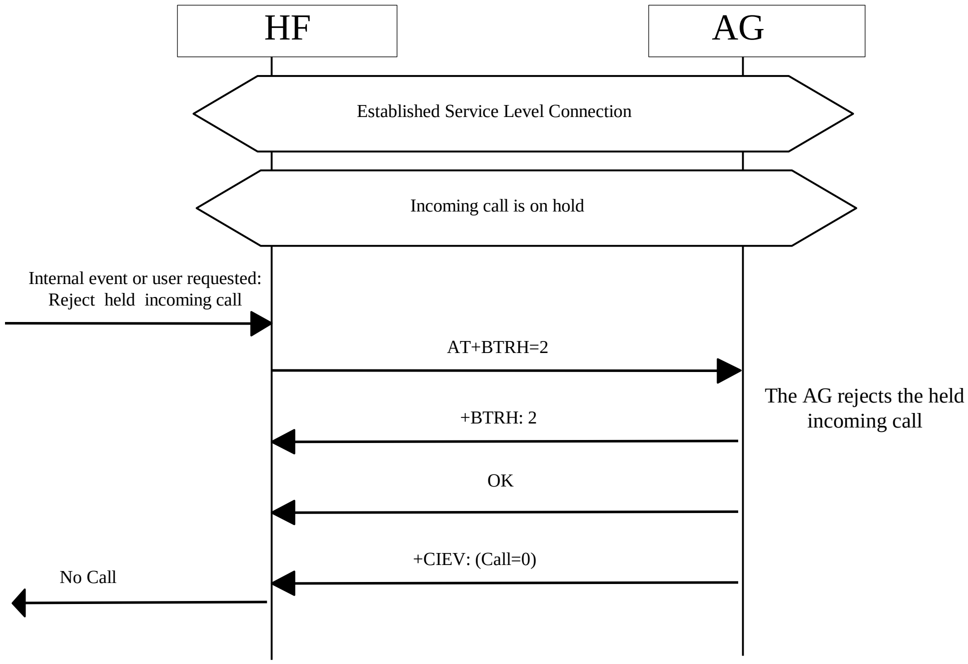

Figure 4.57: Reject a held incoming call from HF

- The user may reject the incoming voice call on hold by using the proper means provided by the HF. Either of the following two sequences shall be permissible by the HF and AG:
  - The HF may send the AT+BTRH command with the parameter <n> set to 2. The AG shall then begin the procedure for rejecting the incoming call that was put on hold. The AG shall send +BTRH Response with the parameter <n> set to 2 to notify the HF that the held incoming call was rejected.
  - The HF may send the AT+CHUP command to reject the held incoming call. The AG shall reject the held call and send the OK indication to the HF.
- The AG shall send the +CIEV Response with the call status set to 0.

4.30.7 Reject a Held Incoming Call from AG

The following additional precondition applies to this procedure:

- An incoming call was put on hold.

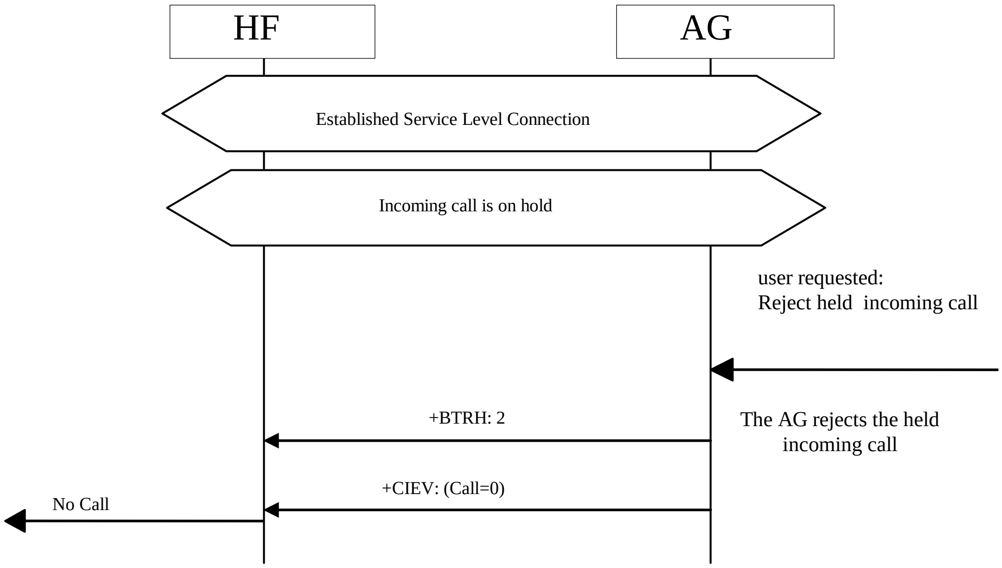

Figure 4.58: Reject a held incoming call from AG

The user may reject the incoming voice call on hold by using the proper means provided by the AG unit. The AG shall then send +BTRH Response with the parameter <n> set to 2 to notify HF that the held incoming call was rejected.

- The AG shall also send the +CIEV Response with the call status parameter set to 0 to indicate that the AG is currently not in a call.

4.30.8 Held Incoming Call Terminated by Caller

The following additional precondition applies to this procedure:

- An incoming call was put on hold.

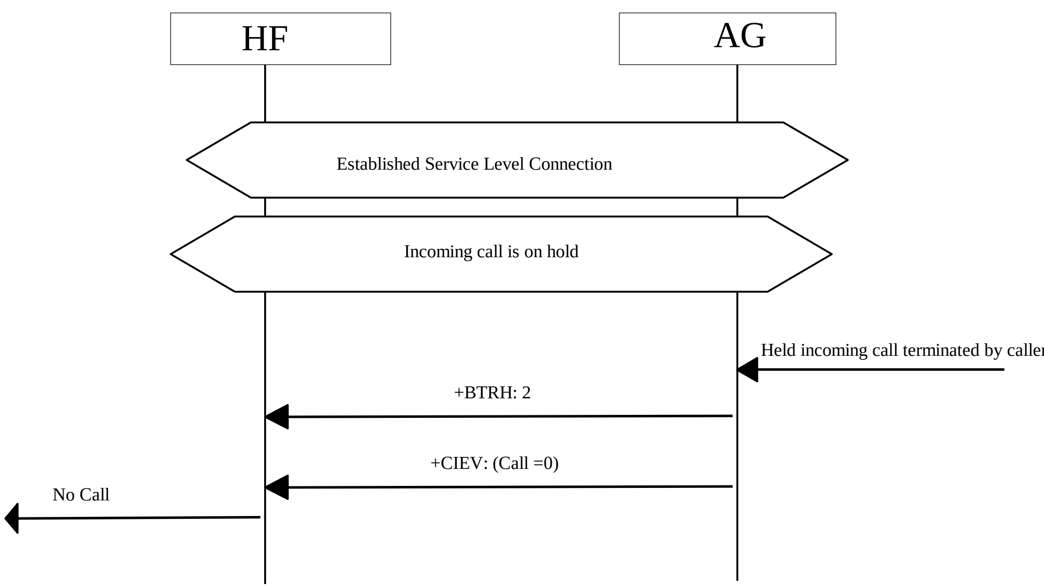

Figure 4.59: Held incoming call terminated by caller

- The caller may terminate the held incoming call. The AG shall then send +BTRH Response with the parameter <n> set to 2 to notify the HF that the held incoming call was terminated.
- The AG shall send the +CIEV Response with the Call status parameter set to 0 to indicate that the AG is currently not in a call.

4.31 Subscriber Number Information

This procedure allows HF to query the AG subscriber number.

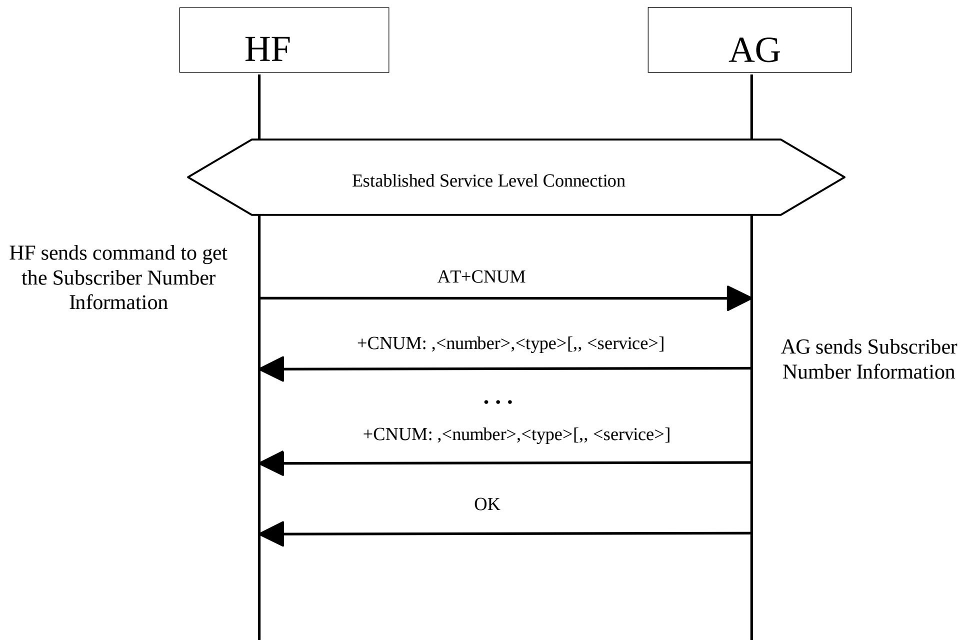

Figure 4.60: Query Subscriber Number Information of AG

This procedure illustrates AG response to the query of an empty subscriber number.

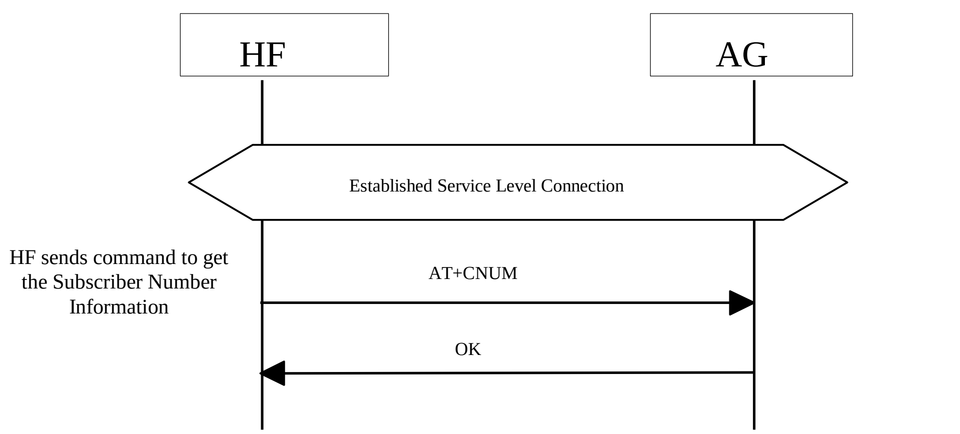

Figure 4.61: Empty Subscriber Number Information from AG

The following precondition applies for this procedure:

- An ongoing Service Level Connection between the HF and AG shall exist. If this connection does not exist, the HF shall establish a connection using the "Service Level Connection set up" procedure described in Section 4.2.
- The HF shall send the AT+CNUM command to query the AG subscriber number information.
- If the subscriber number information is available, the AG shall respond with the +CNUM response. If multiple numbers are available, the AG shall send a separate +CNUM response for each available number.
- The AG shall signal the completion of the AT+CNUM action command with an OK response. The OK will follow zero or more occurrences of the +CNUM response. (See Figure 4.57 and Figure 4.58).

4.32 Enhanced Call Status Mechanisms

4.32.1 Query List of Current Calls in AG

The HF shall execute this procedure to query the list of current calls in AG.

The following precondition applies for this procedure:

- A Service Level Connection exists between the AG and HF devices. If no current Service Level Connection exists, the HF shall first initiate one.

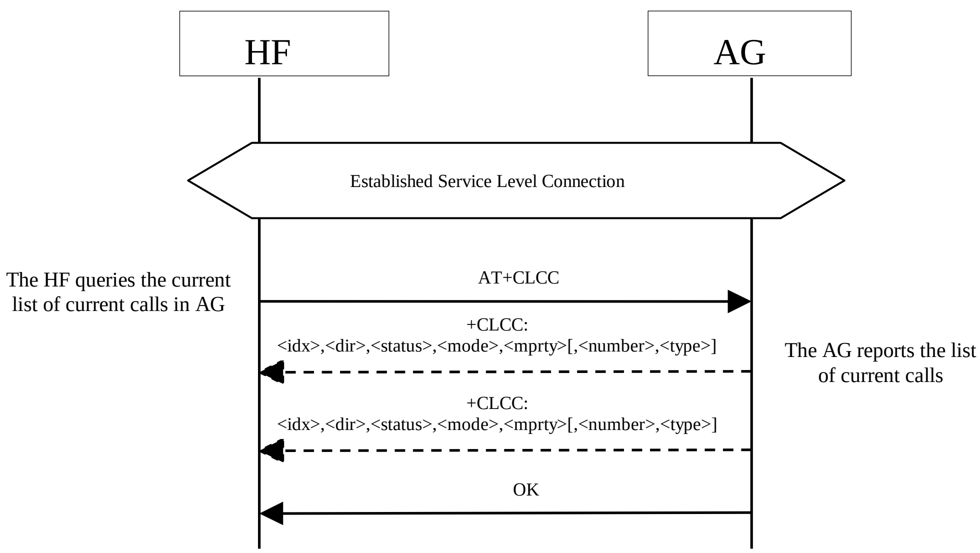

Figure 4.62: Query List of Current Calls

- HF shall find out the list of current calls in AG by sending the AT+CLCC command.
- If the command succeeds and if there is an outgoing (Mobile Originated) or an incoming (Mobile Terminated) call in AG, AG shall send a +CLCC response with appropriate parameters filled in to HF.
- If there are no calls available, no +CLCC response is sent to HF.
- The AG shall always send OK response to HF.

4.33 Enhanced Call Control Mechanisms

As stated earlier, the Enhanced Call Control mechanism is simply an extension of the current AT+CHLD command. These extensions are defined as additional arguments to the AT+CHLD command. The new arguments for this command include an index of a specific call as indicated in the +CLCC response.

4.33.1 Release Specified Call Index

The HF shall execute this procedure to release a specific call in the AG.

The following precondition applies for this procedure:

- A Service Level Connection exists between the AG and HF devices. If no current Service Level Connection exists, the HF shall first initiate one.

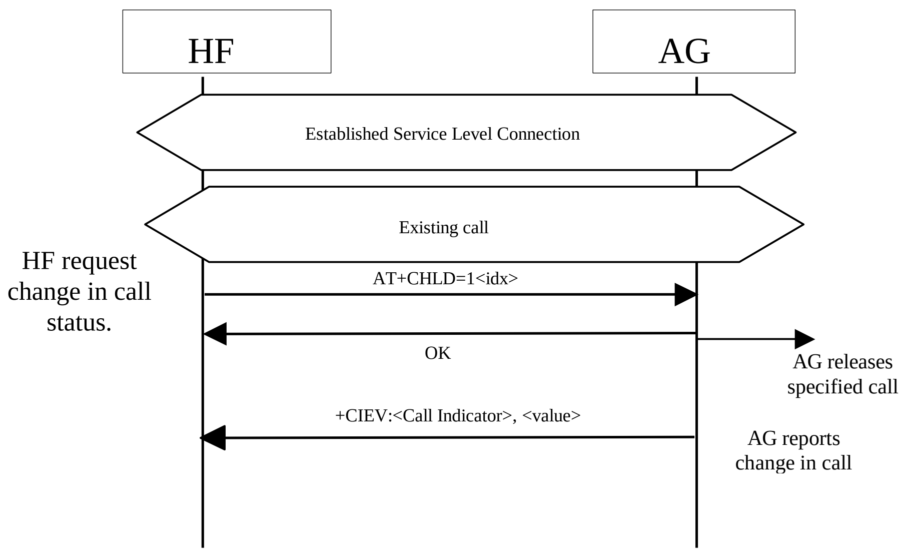

Figure 4.63: Release Specified Active Call

- The HF shall send the AT+CHLD=1<idx> command to release a specific active call.
- The AG shall release the specified call.

If there is a change in the call status, the AG shall report the change in call status. If there is a change in the held call status, the AG shall report the change in call held status.

If the index (<idx>) is not valid, the AG shall report the proper error code.

4.33.2 Private Consultation Mode

The HF shall execute this procedure to place all parties of a multiparty call on hold with the exception of the specified call.

The following precondition applies for this procedure:

- A Service Level Connection exists between the AG and HF devices. If no current Service Level Connection exists, the HF shall first initiate one.
- Existing multiparty call is active in AG.

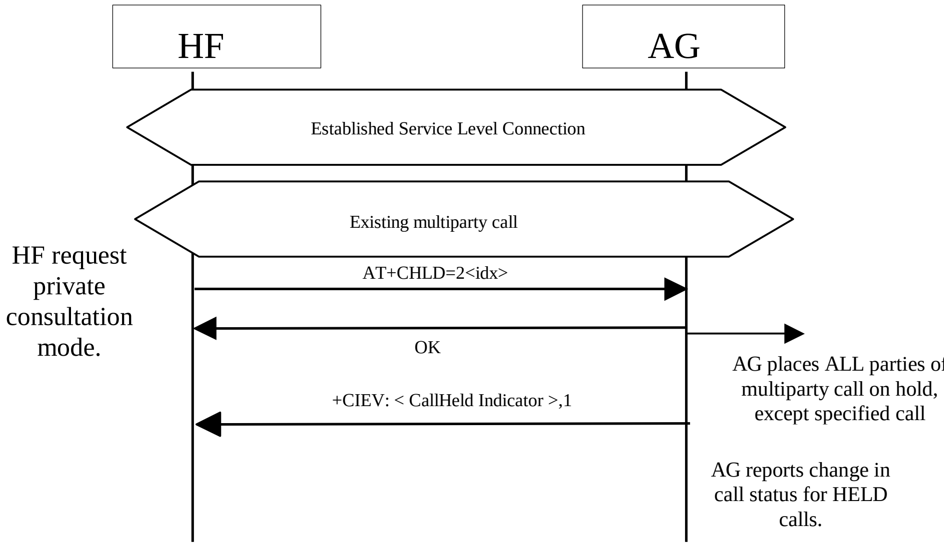

Figure 4.64: Request Private Consultation Mode

- HF shall send the AT+CHLD=2<idx> command to request private consultation mode.
- AG shall place all other parties of call on hold.
- AG shall report the change in status of the held parties.
- If the index (<idx>) is not valid, the AG shall respond with the proper error code.

4.34 Indicators Activation and Deactivation

The HF shall execute this procedure to change the subset of indicators that shall be sent by the AG.

The following preconditions apply for this procedure:

An ongoing Service Level Connection between the AG and the HF shall exist. If this connection does not exist, the HF shall autonomously establish the Service Level Connection using the proper procedure as described in Section 4.2.

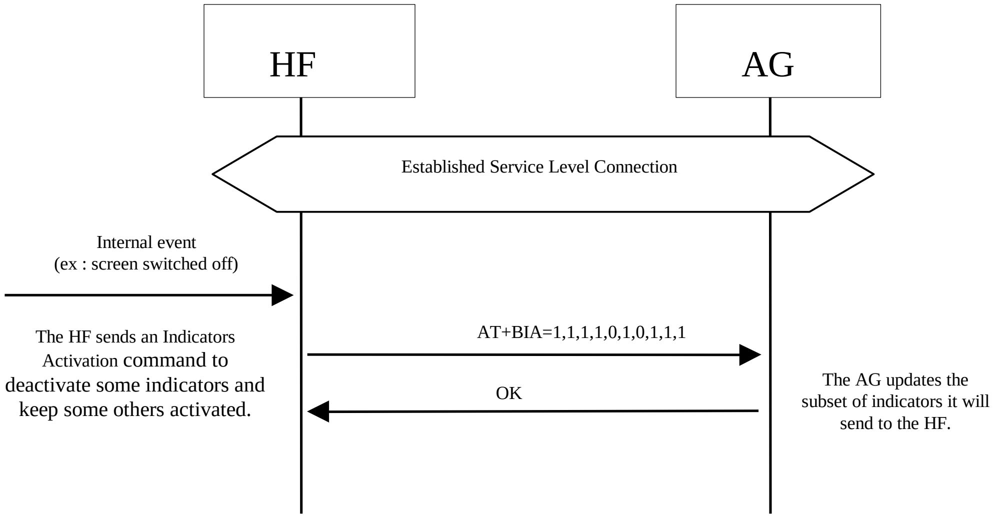

Figure 4.65: Request Activation / Deactivation of Indicators example

Figure 4.65 does not imply a mandatory order for CIND fields, nor does it imply a mandatory set of CIND indicator fields.

The HF shall issue the AT+BIA command if it needs to change the activated/deactivated status of indicators in the AG.

The AG shall send the OK result code to the HF after processing a correctly formatted command.

The AG shall send the ERROR result code to the HF if the command is incorrectly formatted.

Following the successful processing of an AT+BIA command, the AG shall not send the indicators that are deactivated.

The AG shall send the activated indicators response only if the event reporting is enabled.

The effect of the AT+BIA command shall persist during the current Service Level Connection only. When an Service Level Connection is terminated and a new Service Level Connection is established all indicators are activated by default.

It is valid to send the AT+BIA command while the event reporting is disabled. If the event reporting is enabled before the Service Level Connection is terminated, the AG shall send only the indicators that were activated by the most recently processed AT+BIA command.

If the event reporting (CMER) is disabled and then re-enabled, the AG shall send only the indicators that were activated by the most recently processed AT+BIA command, or all indicators in the case where no AT+BIA was sent by the HF.

The AT+BIA command has no impact on the response to the "AT+CIND?" read command (see Section 5.2).

A restriction to the AT+BIA command applies to the indicators call, call status and held call. The AG shall always consider these indicators activated, even if the HF requests their deactivation.

It is mandatory for the AG to support the AT+BIA command.

It is optional for the HF to support the AT+BIA command.

It is optional for HF device to use the AT+BIA command.

See Section 5.2 for more information on the AT+CMER (event reporting) command.

4.35 HF Indicators

The HF Indicator feature is used to allow a Hands-Free device to notify the values of certain HF indicators to an Audio Gateway. The HF may share information such as Battery Level or Enhanced Driver Safety on/off using this feature.

As described in Section 4.2.1, the devices shall determine if the HF Indicators feature is supported by verifying the remote's BRSF bits.

4.35.1 Transfer of HF Supported HF Indicators

During Service Level Connection Establishment procedure (see Section 4.2.1), the HF shall send to the AG a list of supported HF indicators. These shall be represented using the appropriate 16-bit assigned number for each as apportioned within the Assigned Numbers [8] web page on https://www.bluetooth.com/specifications/assigned-numbers/. The HF shall not send any assigned numbers that have not been defined by the Bluetooth SIG.

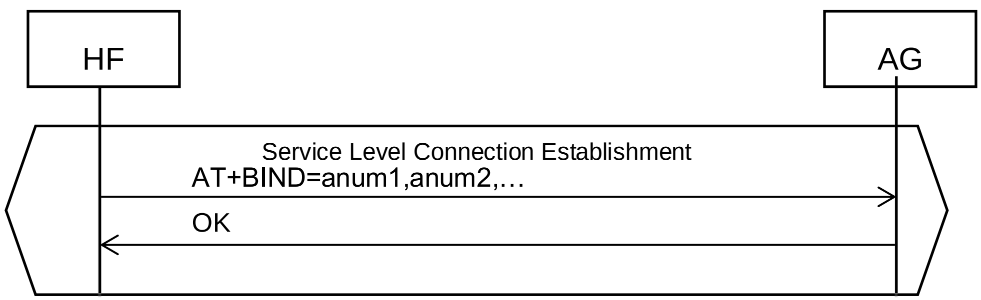

Figure 4.66: HF sends its list of supported HF Indicators

4.35.2 Transfer of the AG Supported HF Indicators

Once the HF has sent its list of supported HF indicators, it shall determine which indicators are supported by the AG. The HF shall send the AT+BIND=? command to the AG. The AG shall respond with +BIND indicating which HF indicator(s) are supported by the AG.

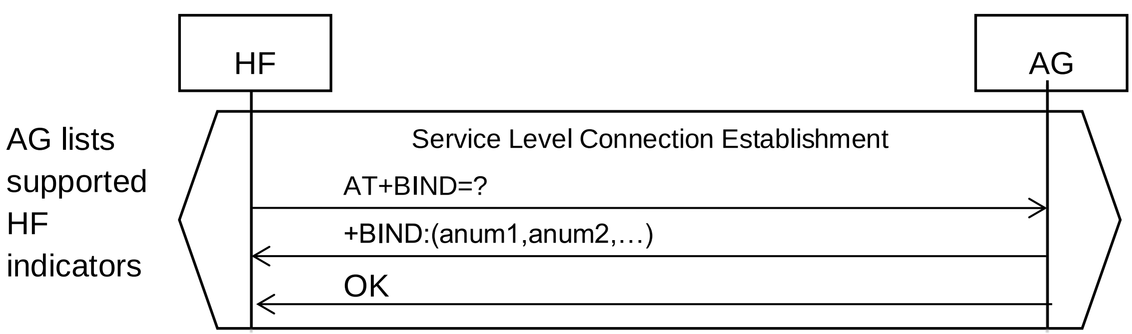

Figure 4.67: AG lists support for HF Indicators

4.35.3 Transfer of Enabled HF Indicators from the AG to the HF

Once the HF has determined which HF Indicators are supported by the AG as described in Section 4.35.2, the HF shall determine which HF Indicators the AG is enabling to receive by sending the AT+BIND? command. The AG shall then send one or more +BIND: anum,state response with the assigned number and state (0= disabled/1= enabled) of each supported HF indicator followed by an OK.

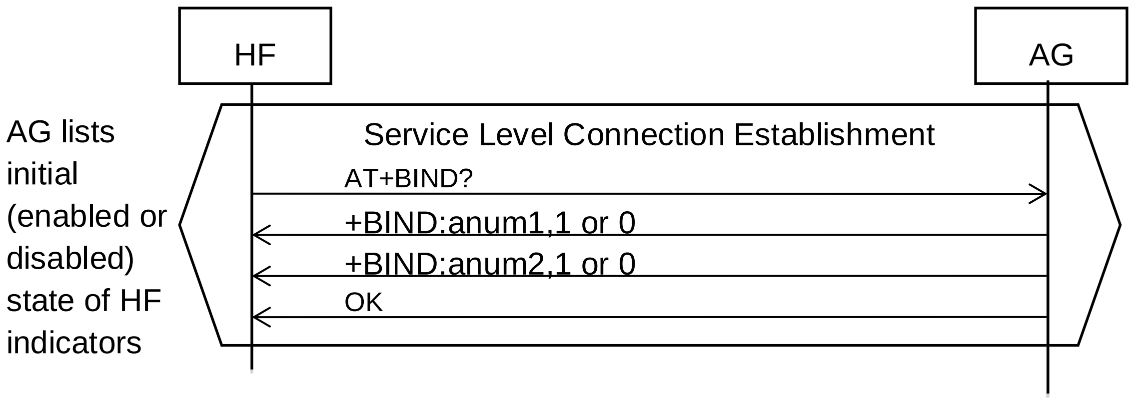

Figure 4.68: AG lists enabled / disabled state of each HF Indicator

4.35.4 Activation / Deactivation of the AG's supported HF Indicators

The AG may change the enabled/disabled state of any of the HF Indicators supported by the HF. To change the state, the AG shall send an unsolicited +BIND: anum,state response code. Whenever the HF receives an unsolicited +BIND indication from the AG that changes the state of a particular HF indicator from disabled to enabled, the HF should send the current state of that indicator to the AG using the +BIEV command (see Section 4.35.5). This synchronizes the states of the HF indicators across the devices.

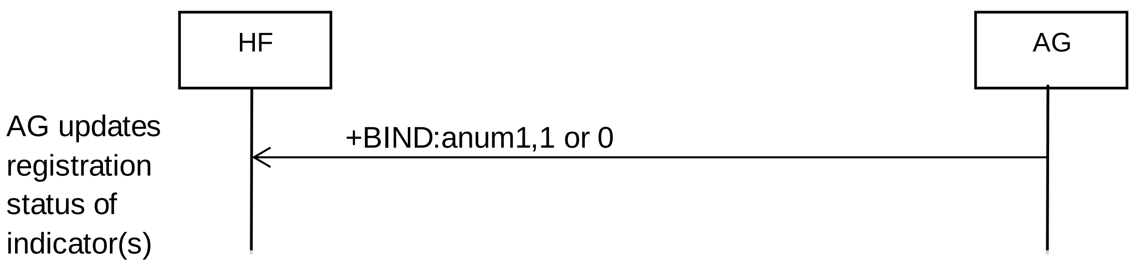

Figure 4.69: HF Indicator is set to enabled or disabled

4.35.5 Transfer of HF Indicator Values

When situational conditions change within the HF, the HF shall notify the AG of the change in value using the AT+BIEV=anum, value command. The AG shall then acknowledge the receipt of the value updates with an OK. If the AG does not support the indicator being reported by the HF or if it is disabled, the AG shall return an ERROR response code.

The assigned number and their associated values are defined by the Bluetooth SIG within the Assigned Numbers [8] web page.

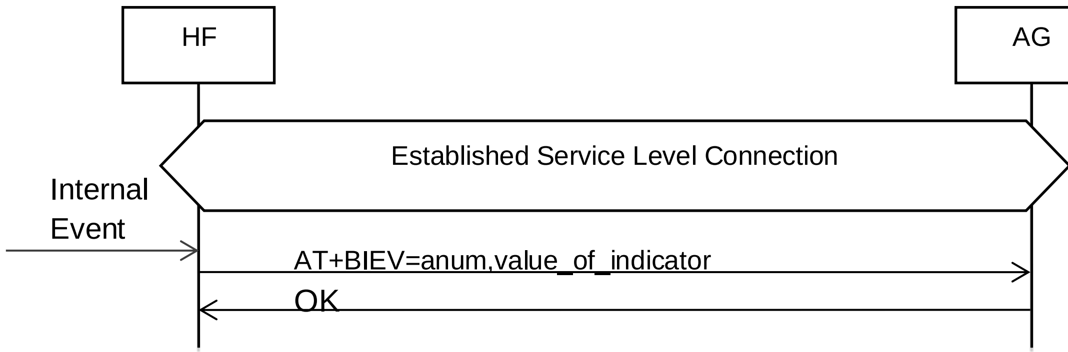

Figure 4.70: Value of HF Indicator is updated

The HF should only provide HF Indicator value updates for those that have been set as enabled by the AG. The AG shall respond with an ERROR response code if it receives updates for disabled or unknown HF indicators or values that are out of bounds.

5 AT Command and Results Codes

5.1 General

For the exchange of the commands and unsolicited results codes, the format, syntax and procedures of 3GPP 27.007 [2] shall be taken as reference. The following rules specifically apply for the HFP specification:

- Only one command (or unsolicited result code) per command line needs to be expected.
- The AG, by default, shall not echo the command characters.
- The AG shall always transmit result codes using verbose format.
- The characters below shall be used for AT commands and result codes formatting:

  <cr> corresponds to the *carriage return (0/13)* as stated in [5].

  <lf> corresponds to the *line feed (0/10)* as stated in [5].
- The format of an AT command from the HF to the AG shall be:

  <AT command><cr>
- The format of the OK code from the AG to the HF shall be:

  <cr><lf>OK<cr><lf>
- The format of the generic ERROR code from the AG to the HF shall be:

  <cr><lf>ERROR<cr><lf>
- The format of an unsolicited result code from the AG to the HF shall be:

  <cr><lf><result code><cr><lf>

The Hands-Free Profile uses a subset of AT commands and result codes from existing standards; these are listed in Section 5.2. Section 5.3 lists the new Bluetooth defined AT commands and result codes not re-used from any existing standard.

In general, the AG shall use the OK code, as described in Section 5.2, for acknowledgement of the proper execution of a command and respond with the proper error indication to any unknown command received from the HF.

It is mandatory for the AG to properly respond to any error condition and for the HF to properly process the corresponding error indication code received from the AG. The code ERROR, as described in Section 5.2, shall be used as error indication for this purpose.

The HF shall always ignore any unknown or unexpected indication code received from the AG for AT commands defined in this specification. Handling of AT commands not defined in this specification is left to the implementation. The only exception is the case in which the AG issues a “Mobile Equipment Error” indication using the +CME ERROR: result code (see [2]). In this case, the HF shall interpret this result code in the same way as if it was a generic ERROR code.

As a general rule, when an AT command or result code of this specification is implemented, support for the associated parameters described in this specification, and all their corresponding possible values, shall be considered mandatory unless otherwise explicitly stated in each particular case.

5.2 AT Capabilities Re-Used from GSM 07.07 and 3GPP 27.007

The re-used AT commands and unsolicited result codes for implementing the functionality described in this specification are listed below:

- ATA

  Standard call answer AT command. See Annex G in [2].

- ATDdd…dd;

  Standard AT command intended for placing a call to a phone number. Only voice calls are covered in this specification. See Section 6.2 in [2].

- ATD>nnn...;

  Extension of the standard ATD command, intended for memory dialing. Only voice calls are covered in this specification. See Section 6.3 in [2].

- ERROR

  Standard error indication code. It shall be issued on detection of any syntax, format or procedure error condition. The "Mobile Equipment Error" report code "+CME ERROR:" is covered below. See Annex B in [2].

- OK

  Standard acknowledgement to the execution of a command. See Annex B in [2].

- NO CARRIER, BUSY, NO ANSWER, DELAYED, REMOVED FROM THE NETWORK

  Extended response indication codes for AT commands. These codes shall be issued from the AG to the HF as responses to AT commands from the HF to the AG or from the AG as unsolicited result codes. These are in addition to the +CME ERROR: responses.

- RING

  Standard "incoming call" indication. See Annex B in [2].

- AT+CCWA

  Standard "Call Waiting notification" AT command. Within the AT+CCWA=[<n>[,<mode>[,<class>]]]command, only enabling/disabling of the Call Waiting notification unsolicited result code +CCWA , using the <n> parameter, is covered in this specification. See Section 7.12 in [2].

- +CCWA

  Standard "Call Waiting notification" unsolicited result code.

  In the +CCWA result code only <number> and <type> parameters are covered in this specification. Other parameters are not considered relevant in this specification and shall be ignored by the HF.

  The <number> parameter shall be a text string and shall always be contained within double-quotes.

  The <type> field specifies the format of the phone number provided, and can be one of the following values:

values 128-143: The phone number format may be a national or international format, and may contain prefix and/or escape digits. No changes on the number presentation are required.

values 144-159: The phone number format is an international number, including the country code prefix. If the plus sign ("+") is not included as part of the number and shall be added by the AG as needed.

values 160-175: National number. No prefix nor escape digits included.

See Section 7.12 in [2].

AT+CHLD

Standard call hold and multiparty handling AT command. In the AT+CHLD=<n> command, this specification only covers values for <n> of 0, 1, 1<idx>, 2, 2<idx>, 3 and 4, where:

- 0 = Releases all held calls or sets User Determined User Busy (UDUB) for a waiting call.
- 1 = Releases all active calls (if any exist) and accepts the other (held or waiting) call.
- 1<idx> = Releases call with specified index (<idx>).
- 2 = Places all active calls (if any exist) on hold and accepts the other (held or waiting) call.
- 2<idx> = Request private consultation mode with specified call (<idx>). (Place all calls on hold EXCEPT the call indicated by <idx>.)
- 3 = Adds a held call to the conversation.
- 4 = Connects the two calls and disconnects the subscriber from both calls (Explicit Call Transfer). Support for this value and its associated functionality is optional for the HF.
- Where both a held and a waiting call exist, the above procedures shall apply to the waiting call (i.e., not to the held call) in conflicting situation.

The test command AT+CHLD=? may be used for retrieving information about the call hold and multiparty services available in the AG (see Section 4.2.1).

See Section 7.13 in [2] and Section 4.5.5.1 in [6] for details.

• AT+CHUP

Standard hang-up AT command. Execution command causes the AG to terminate the currently active call. This command shall have no impact on the state of a held call except in the use of rejecting a call placed on hold by the Respond and Hold feature as defined in Section 4.30.6.

See Section 6.5 in [2].

AT+CHUP is also used as the command to reject any incoming call prior to answer.

• AT+CIND

Standard indicator update AT command. Only read command AT+CIND? and test command AT+CIND=? are required in this specification.

The AT+CIND? read command is used to get current status of the AG indicators.

The AG shall return all the indicators listed in the AT+CIND=? command.

The deactivation of any indicator(s) using AT+BIA command shall have no effect on the AG's response to the AT+CIND? read command.

The AT+CIND=? test command is used to retrieve the mapping between each indicator supported by the AG and its corresponding range and order index. It shall be issued at least once before any other command related to these indicators (AT+CIND? or AT+CMER) is used.

The Hands Free Profile specification limits the number of indicators returned by the AG to a maximum of 20.

The following indicators are covered in this specification:

- service: Service availability indication, where:

  <value>=0 implies no service. No Home/Roam network available.

  <value>=1 implies presence of service. Home/Roam network available.

- call: Standard call status indicator, where:

  <value>=0 means there are no calls in progress

  <value>=1 means at least one call is in progress

- callsetup: Bluetooth proprietary call set up status indicator[4]. Support for this indicator is optional for the HF. When supported, this indicator shall be used in conjunction with, and as an extension of the standard call indicator. Possible values are as follows:

  <value>=0 means not currently in call set up.

  <value>=1 means an incoming call process ongoing.

  <value>=2 means an outgoing call set up is ongoing.

  <value>=3 means remote party being alerted in an outgoing call.

See Section 8.9 in [2].

- callheld: Bluetooth proprietary call hold status indicator. Support for this indicator is mandatory for the AG, optional for the HF. Possible values are as follows:

  0= No calls held

  1= Call is placed on hold or active/held calls swapped

  (The AG has both an active AND a held call)

  2= Call on hold, no active call

- signal: Signal Strength indicator, where:

  <value>= ranges from 0 to 5

- roam: Roaming status indicator, where:

  <value>=0 means roaming is not active

[4] This status indicator is not defined in the GSM 07.07 specification

<value>=1 means a roaming is active

- battchg: Battery Charge indicator of AG, where:

  <value>=ranges from 0 to 5

- +CIND

  Standard list of current phone indicators. See Section 8.9 in [2].

- AT+CLCC

  Standard list current calls command. See Section 7.18 in [2].

- +CLCC

  Standard list current calls result code. See Section 7.18 in [2].

  Supported parameters are as follows:

  - idx= The numbering (starting with 1) of the call given by the sequence of setting up or receiving the calls (active, held or waiting) as seen by the served subscriber. Calls hold their number until they are released. New calls take the lowest available number.
  - dir= 0 (outgoing), 1 (incoming)
  - status=
    - 0 = Active
    - 1 = Held
    - 2 = Dialing (outgoing calls only)
    - 3 = Alerting (outgoing calls only)
    - 4 = Incoming (incoming calls only)
    - 5 = Waiting (incoming calls only)
    - 6 = Call held by Response and Hold
  - mode= 0 (Voice), 1 (Data), 2 (FAX)
  - mpty=
    - 0 - this call is NOT a member of a multi-party (conference) call
    - 1 - this call IS a member of a multi-party (conference) call
  - number (optional) = The <number> parameter shall be a text string and shall always be contained within double quotes.
  - type (optional) = The <type> parameter specifies the format of the phone number provided. The <type> field can be one of the following values:

    - values 128 to 143: The phone number format may be a national or international format and may contain prefix and/or escape digits. No changes to the number presentation are required.

    - values 144 to 159: The phone number format is an international number format, including the country code prefix. If the plus sign (“+”) is not included as part of the number, the plus sign (“+”) shall be added by the AG as needed.

- values 160 to 175: The phone number format is a national number format. No prefix nor escape digits are included.

If the <number> parameter is not provided, the <type> parameter shall be excluded.

- **AT+COPS**

  The AT+COPS=3,0 shall be sent by the HF to the AG prior to sending the AT+COPS? command. AT+COPS=3,0 sets the format of the network operator string to the long format alphanumeric.

  The AT+COPS? command is used for reading network operator. This profile shall only support the "reading" of the name of the network operator. The response to this command from the AG shall return a +COPS:<mode>,<format>,<operator> where:

  <mode> contains the current mode and provides no information with regard to the name of the operator.

  <format> specifies the format of the <operator> parameter string, and shall always be 0 for this specification.

  <operator> specifies a quoted string in alphanumeric format representing the name of the network operator. This string shall not exceed 16 characters. See Section 7.3 in [2].

- AT+CMEE

  Standard AT command used to enable the use of result code +CME ERROR: <err> as an indication of an error relating to the functionality of the AG.

  The set command AT+CMEE=1 is covered in this specification.

- +CME ERROR

  This is the Extended Audio Gateway Error Result Code response. Format of the response is: +CME ERROR: <err>. The format of <err> shall be numeric in this specification. The possible values for <err> covered in this specification are described below. These error codes may be provided instead of the standard ERROR response code to provide additional information to the HF. The ERROR response code is still allowed while using the Extended Audio Gateway Error Result Codes.

  +CME ERROR: 0 – AG failure

  +CME ERROR: 1 – no connection to phone

  +CME ERROR: 3 – operation not allowed

  +CME ERROR: 4 – operation not supported

  +CME ERROR: 5 – PH-SIM PIN required

  +CME ERROR: 10 – SIM not inserted

  +CME ERROR: 11 – SIM PIN required

  +CME ERROR: 12 – SIM PUK required

  +CME ERROR: 13 – SIM failure

  +CME ERROR: 14 – SIM busy

+CME ERROR: 16 – incorrect password

+CME ERROR: 17 – SIM PIN2 required

+CME ERROR: 18 – SIM PUK2 required

+CME ERROR: 20 – memory full

+CME ERROR: 21 – invalid index

+CME ERROR: 23 – memory failure

+CME ERROR: 24 – text string too long

+CME ERROR: 25 – invalid characters in text string

+CME ERROR: 26 – dial string too long

+CME ERROR: 27 – invalid characters in dial string

+CME ERROR: 30 – no network service

+CME ERROR: 31 - network Timeout.

+CME ERROR: 32 – network not allowed – Emergency calls only

- AT+CLIP

  Standard "Calling Line Identification notification" activation AT command. It enables/disables the Calling Line Identification notification unsolicited result code +CLIP. See Section 7.6 in [2].

- +CLIP

  Standard "Calling Line Identification notification" unsolicited result code.

  In the +CLIP: <number>, type> [,<subaddr>,<satype> [,[<alpha>] [,<CLI validity>]]] result code. Only <number> and <type> parameters are covered in this specification. Other parameters are not considered relevant in this specification and shall be ignored by the HF.

  The <number> parameter shall be a text string and shall always be contained within double-quotes.

  The <type> field specifies the format of the phone number provided, and can be one of the following values:

  - values 128-143: The phone number format may be a national or international format, and may contain prefix and/or escape digits. No changes on the number presentation are required.
  - values 144-159: The phone number format is an international number, including the country code prefix. If the plus sign ("+") is not included as part of the number and shall be added by the AG as needed.
  - values 160-175: National number. No prefix nor escape digits included.
  - See Section 7.11 in [2].

- AT+CMER

  Standard event reporting activation/deactivation AT command.

In the AT+CMER=[<mode>[,<keyp>[,<disp>[,<ind> [,<bfr>]]]]] command, only the <mode>, and <ind> parameters are relevant for this specification. Only their values <mode>=(3) and <ind>=(0,1) are covered in this specification. See Section 8.10 in [2].

The following examples show how the AT+CMER command may be used for activating or deactivating the “indicator events reporting” result code:

AT+CMER=3,0,0,1 activates “indicator events reporting”.

AT+CMER=3,0,0,0 deactivates “indicator events reporting”.

- +CIEV

  Standard “indicator events reporting” unsolicited result code.

  In the +CIEV: <ind>,<value> result code, only the indicators stated in the AT+CIND command above are relevant for this specification where:

  - <ind>: Order index of the indicator within the list retrieved from the AG with the AT+CIND=? command. The first element of the list shall have <ind>=1.
  - <value>: current status of the indicator.

  If the HF receives any unknown indicator or value, it shall ignore it.

  See Section 8.10 in [2].

- AT+VTS

  Standard DTMF generation AT command. Only the AT+VTS=<DTMF> command format is covered in this specification.

  See Annex C.2.11 in [2].

- AT+CNUM

  Syntax:

AT+CNUM	(Retrieve Subscriber Number Information)
AT+CNUM=?	(Test Subscriber Number Information – Not Implemented)

  Description:

  Command issued by HF for the “Subscriber Number Information” feature in the AG.

  Only the action command AT+CNUM format is used.

- **+CNUM**

  Syntax:

  +CNUM: [<alpha>],<number>, <type>,[<speed> ,<service>] (Response for AT+CNUM)

  Description:

  Standard Response used for sending the “Subscriber Number Information” from AG to HF.

  The AG shall send the +CNUM: response for the AT+CNUM from the HF.

  Values:

  - <alpha>: This optional field is not supported, and shall be left blank.

- <number>: Quoted string containing the phone number in the format specified by <type>.
- <type> field specifies the format of the phone number provided, and can be one of the following values:
- values 128-143: The phone number format may be a national or international format, and may contain prefix and/or escape digits. No changes on the number presentation are required.
- values 144-159: The phone number format is an international number, including the country code prefix. If the plus sign ("+") is not included as part of the number and shall be added by the AG as needed.
- values 160-175: National number. No prefix nor escape digits included.
- <speed>: This optional field is not supported, and shall be left blank.
- <service>: Indicates which service this phone number relates to. Shall be either 4 (voice) or 5 (fax).

Example:

+CNUM: ,"5551212",129,,4

See section 7.1 in [2].

5.3 Bluetooth Defined AT Capabilities

The GSM 07.07 [2] format and syntax rules shall be taken as the reference for these commands.

The new Bluetooth specific AT capabilities are listed below:

- **AT+BIA** (Bluetooth Indicators Activation)

  Syntax: AT+BIA=[[<indrep 1>][,[<indrep 2>][,…[,[<indrep n>]]]]]

  Description:

  Activates or deactivates the indicators individually. The mapping of the indicators is given by the “AT+CIND=?” test command (see Section 5.2).

  <*indrep x*>: reporting state of the indicator x. 1 to activate, 0 to deactivate.

  If an indicator state is omitted between commas, the current reporting state of that indicator shall not change. For example, if the HF sends the command “AT+BIA=,1,,0” only the second and fourth indicators may be affected. The reporting state of indicators one and three shall remain unchanged.

  If the AG supports more indicators than the number of indicator-reporting-states provided by the HF, the AG shall maintain the current reporting states of those indicators. For example, if the AG supports five indicators and the HF sends the command “AT+BIA=1,0,1“ then only the first three AG indicators may be affected by the command.

  Call, Call Setup, and Held Call indicators have been defined as mandatory indicators. This implies that whatever the reporting state the HF gives, these indicators shall always been kept activated by the AG.

  The AG shall gracefully ignore any excess parameter(s) at the end.

  The AG shall silently ignore a request to deactivate a mandatory indicator.

The previous three points allow the HF to activate or deactivate all the indicators, except the mandatory ones, by using a fixed string.

For example, if the maximum indicator count is 20:

All indicators can be set to active by using the fixed string:

AT+BIA=1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1

and to inactive (except for the always active ones) by using:

AT+BIA=0,0,0,0,0,0,0,0,0,0,0,0,0,0,0,0,0,0,0,0

The actual number of allowed indicators is defined by the AT+CIND command.

- **AT+BINP (Bluetooth INPut)**

  Syntax: AT+BINP=<datarequest>

  Expected response: +BINP: <dataresp$_1$>…<dataresp$_n$>

  Description:

  Command used for requesting some specific data input from the AG[5]. On reception of this command the AG shall perform the proper actions such that the requested information is sent back to the HF using the +BINP response.

  The type of data the HF shall expect in the <dataresp> parameter returned by the AG depends on the information requested in each case.

  Only support for execution command is mandated. Neither the read nor test commands are mandatory.

  Values:

  <datarequest>: 1, where

  1 = Phone number corresponding to the last voice tag recorded in the HF.

  <dataresp1..n>: Data parameters returned by the AG. Their contents depend on the value of the <datarequest> parameter as follows:

<datarequest> value	<dataresp> parameters
1	<Phone number>: Phone number string (max. 32 digits). The format (type of address) of the phone number string shall conform with the rules stated in [5], sub-clause 10.5.4.7, for a value (in integer format) of the type of address octet of 145, if dialing string includes international access code character “+”, and for a value of 129 otherwise.

- **AT+BLDN** (Bluetooth Last Dialed Number)

  Syntax: AT+BLDN

  Description:

[5] AT+BINP was created with future extensibility in mind. While the Hands-Free Profile only specifies a <datarequest> value of 1 (i.e., phone number), future profiles may choose to add values for <datarequest> to support the retrieval of additional data from the AG.

Command used for calling the last phone number dialed. On reception of this command, the AG shall set up a voice call to the last phone number dialed.

Only support for execution command is mandated. Neither the read nor test commands are mandatory.

- **AT+BVRA** (Bluetooth Voice Recognition Activation)

  Syntax: AT+BVRA=<vrec>

  Description:

  Enables/disables the voice recognition function in the AG. If the Enhanced Voice Recognition Status feature is supported, this command is used to indicate to the AG that the HF is ready to render audio output.

  Only support for execution command is mandated. Neither the read nor test commands are mandatory. If the Enhanced Voice Recognition Status feature is supported, the AG shall only transmit audio to the HF if the AG has previously received AT+BVRA=2 from the HF and an eSCO link has been established.

  The value 2 shall only be used if both the AG and the HF support the Enhanced Voice Recognition Status feature.

  Values:

  <vrec>: 0, 1, or 2, entered as integer values, where

  0 =

  - Disable Voice recognition in the AG.
  - If the Enhanced Voice Recognition Status feature is supported, this value terminates the current VR session.
  - If the Enhanced Voice Recognition Status feature is supported, this value may also be used to refuse the +BVRA=1 request from the AG to start a new VR session.

  1 = Enable voice recognition in the AG.

  2 = This value shall only be used if both the AG and the HF support the Enhanced Voice Recognition Status feature. This value indicates that the HF is ready to accept audio when the Audio Connection is first established. The HF shall only send this value if the eSCO link has been established.

  The HF may send this value during an ongoing VR session to terminate audio output from the AG (if there is any) and prepare the AG for new audio input.

- **+BVRA** (Bluetooth Voice Recognition Activation)

  Syntax: +BVRA: <vrect>[,<vrecstate>][,<textualRepresentation>]

  Description:

  Unsolicited result code used to notify the HF when the voice recognition function in the AG is activated/deactivated autonomously from the AG. If the Enhanced Voice Recognition Status feature is supported, then this value is also used to indicate state changes in the AG´s voice recognition engine.

The unsolicited +BVRA: 1 result code shall not be sent by the AG to the HF if the corresponding voice recognition activation has been initiated by the HF. Likewise, the unsolicited +BVRA: 0 result code shall not be sent by the AG to the HF if the corresponding voice recognition deactivation has been initiated by the HF, regardless of which side initiated the voice recognition activation.

If the Enhanced Voice Recognition Status feature is supported, the AG shall only transmit audio to the HF if the AG has previously received AT+BVRA=2 from the HF and an eSCO link has been established.

Values:

<vrect>: 0, entered as integer value, where

0 = Voice recognition is disabled in the AG

1 = Voice recognition is enabled in the AG

<vrecstate>:

Bitmask that reflects the current state of the voice recognition engine on the AG. When there is no active voice recognition session, Bit 0 shall be set to 0. If Bit 0 is set to 0, all other bits shall be set to 0.

The <vrecstate> shall be present if both the AG and HF support the Enhanced Voice Recognition Status feature; otherwise, it shall not be present.

Bit	Description
0	If value of this bit is 1, the AG is ready to accept audio input
1	If value of this bit is 1, the AG is sending audio to the HF
2	If value of this bit is 1, the AG is processing the audio input

<textualRepresentation>: <textID>,<textType>,<textOperation>,<string>

The textualRepresentation shall only be present if both the AG and HF support the Voice Recognition Text feature.

<textID>: Unique ID of the current text as a hexadecimal string (a maximum of 4 characters in length, but less than 4 characters in length is valid). TextID shall change if the textType has changed. Each textID value shall be unique for a VR session. TextID values shall not be valid across VR sessions.

<textType>: ID of the textType from the following list:

ID	Description
0	Text recognized by the AG from the audio input provided by the HF
1	Text of the audio output from the AG
2	Text of the audio output from the AG that contains a question
3	Text of the audio output from the AG that contains an error description
4–31	Reserved for future use

<textOperation>: ID of the operation of the text

ID	Description
0	Reserved for future use
1	NewText: Indicates that a new text started. Shall be used when the textID changes
2	Replace: Replace any existing text with the same textID and same text Type
3	Append: Attach new text to existing text and keep the same textID and same text Type
4–31	Reserved for future use

<string>: The <string> parameter shall be a UTF-8 text string and shall always be contained within double quotes. The length of the string depends on the implementation.

The following example of a +BVRA command contains a textual representation of the sentence "Message to Melissa." In the example, the state of the voice recognition engine accepts further audio input, the textID is 12AB, the textType is 'recognized text from audio input' and the operation is 'NewText':

+BVRA: 1,1,12AB,0,1,"Message to Melissa"

- **AT+BRSF** (Bluetooth Retrieve Supported Features)

  Syntax: AT+BRSF=<HF supported features bitmap>

  Description:

  Notifies the AG of the supported features available in the HF, and requests information about the supported features in the AG. The supported features shall be represented as a decimal value.

  Values:

  <HF supported features bitmap>: a decimal numeric string, which represents the value of a 32 bit unsigned integer. The 32 bit unsigned integer represents a bitmap of the supported features in the HF as follows:

Bit	Feature
0	EC and/or NR function
1	Three-way calling
2	CLI presentation capability
3	Voice recognition activation
4	Remote audio volume control
5	Enhanced call status
6	Enhanced call control
7	Codec Negotiation
8	HF Indicators
9	eSCO S4 Settings Supported
10	Enhanced Voice Recognition Status

Bit	Feature
11	Voice Recognition Text
12-31	Reserved for future use

The reserved bits (12-31) shall be initialized to Zero.

- **+BRSF** (Bluetooth Retrieve Supported Features)

Syntax: +BRSF: <AG supported features bitmap>

Description:

Result code sent by the AG in response to the AT+BRSF command, used to notify the HF what features are supported in the AG. The supported features shall be represented as a decimal value.

Values:

<AG supported features bitmap>: a decimal numeric string, which represents the value of a 32 bit unsigned integer. The 32 bit unsigned integer represents a bitmap of the supported features in the AG as follows:

Bit	Feature
0	Three-way calling
1	EC and/or NR function
2	Voice recognition function
3	In-band ring tone capability
4	Attach a phone number for a voice tag
5	Ability to reject a call
6	Enhanced call status
7	Enhanced call control
8	Extended Error Result Codes
9	Codec Negotiation
10	HF Indicators
11	eSCO S4 Settings Supported
12	Enhanced Voice Recognition Status
13	Voice Recognition Text
14-31	Reserved for future use

The reserved bits (14-31) shall be initialized to Zero.

- **AT+NREC** (Noise Reduction and Echo Canceling)

  Syntax: AT+NREC=<nrec>

  Description:

  Command issued to disable any Echo Canceling and Noise Reduction functions embedded in the AG.

  Only support for execution command is mandated. Neither the read nor test commands are mandatory.

  Values:

  <nrec>: 0, entered as integer value, where

  0 = Disable EC/NR in the AG

- **AT+VGM** (Gain of Microphone)

  Syntax: AT+VGM=<gain>

  Description:

  Command issued by the HF to report its current microphone gain level setting to the AG. <gain> is a decimal numeric constant, relating to a particular (implementation dependent) volume level controlled by the HF. This command does not change the microphone gain of the AG; it simply indicates the current value of the microphone gain in the HF.

  Only support for execution command is mandated. Neither the read nor test commands are mandatory.

  Values:

  <gain>: 0 -15, entered as integer values, where

  0 = Minimum gain

  15 = Maximum gain

- **AT+VGS** (Gain of Speaker)

  Syntax: AT+VGS=<gain>

  Description:

  Command issued by the HF to report its current speaker gain level setting to the AG. <gain> is a decimal numeric constant, relating to a particular (implementation dependent) volume level controlled by the HF. This command does not change the speaker gain of the AG; it simply indicates the current value of the speaker volume in the HF.

  Only support for execution command is mandated. Neither the read nor test commands are mandatory.

  Values:

  <gain>: 0 -15, entered as integer values, where

  0 = Minimum gain

15 = Maximum gain

- **+VGM** (Gain of Microphone)

  Syntax: +VGM:<gain>

  Description:

  Unsolicited result code issued by the AG to set the microphone gain of the HF. <gain> is a decimal numeric constant, relating to a particular (implementation dependent) volume level controlled by the HF.

  Due to the small inconsistency between the GSM standard [2]) and the current Headset Profile specification ([3]), the HF shall also accept the "=" symbol, in place of ":", as a valid separator for this unsolicited result code.

  Values:

  <gain>: 0 -15, integer values, where

  0 = Minimum gain

  15 = Maximum gain

- **+VGS** (Gain of Speaker)

  Syntax: +VGS:<gain>

  Description:

  Unsolicited result code issued by the AG to set the speaker gain of the HF. <gain> is a decimal numeric constant, relating to a particular (implementation dependent) volume level controlled by the HF.

  Due to the small inconsistency between the GSM 07.07 standard ([2]) and the current Headset Profile specification [3]), the HF shall also accept the "=" symbol, in place of ":", as valid separator for this unsolicited result code.

  Values:

  <gain>: 0 -15, integer values, where

  0 = Minimum gain

  15 = Maximum gain

- **+BSIR** (Bluetooth Setting of In-band Ring tone)

  Syntax: +BSIR: <bsir>

  Description:

  Unsolicited result code issued by the AG to indicate to the HF that the in-band ring tone setting has been locally changed. The HF may react accordingly by changing its own alert method.

  Values:

  <bsir>: 0 = the AG provides no in-band ring tone

  1 = the AG provides an in-band ring tone

- **AT+BTRH** (Bluetooth Response and Hold Feature)

  Syntax:

  AT+BTRH=<n> (Set command)

  AT+BTRH? (Read Current Status)

  Description:

  Command issued by the HF for the “Response and Hold” feature in the AG.

  This specification defines the use of the set and read command. The AT+BTRH? command shall be used by the HF to query the current “Response and Hold” state of the AG.

  Values:

  <n>: 0, 1, 2 entered as integer values, where

  0 = Put Incoming call on hold

  1 = Accept a held incoming call

  2 = Reject a held incoming call

- **+BTRH** (Bluetooth Response and Hold Feature)

  Syntax: +BTRH: <n> (Response for AT+BTRH)

  Description:

  Result code used to notify the HF whenever the incoming call is either put on hold or accepted or rejected. The AG shall also respond back with this response for the AT+BTRH? command from the HF.

  Values:

  <n>: 0,1,2 entered as integer value, where

  0 = Incoming call is put on hold in the AG

  1 = Held incoming call is accepted in the AG

  2 = Held incoming call is rejected in the AG

- **AT+BCC** (Bluetooth Codec Connection)

  Syntax: AT+BCC

  Description:

  This command is used by the HF to request the AG to start the Codec Connection procedure.

- **AT+BCS** (Bluetooth Codec Selection)

  Syntax: AT+BCS= <u> (u is a Codec ID)

  Description:

  This command confirms the codec to the remote device (AG), and implicitly also which synchronization protocol, will be used on the synchronous connection.

  If no value is included, the command is invalid.

Values:

<u>: All possible Codec IDs, see definition of AT+BAC.

- **+BCS** (Bluetooth Codec Selection)

  Syntax: +BCS: <u> (u is a codec ID)

  Description:

  This command informs the codec to the remote device (HF), and implicitly also which synchronization protocol, will be used on the synchronous connection.

  Values:

  <u>: All possible Codec IDs, see definition of AT+BAC.

- **AT+BAC** (Bluetooth Available Codecs)

  Syntax: AT+BAC= [<u1>[,<u2>[,...[,<un>]]]] (u1,u2, ..., un are a codec IDs)

  Description:

  This command informs the remote device (AG) about what codecs (see Table 3.3) the HF supports.

  The Codec ID for the mandatory narrow band codec (CVSD) shall always be included.

  If Wide Band Speech is supported, then the mandatory codec (mSBC) shall be included unless it is temporarily unavailable.

  If Super Wide Band Speech is supported, then the mandatory codec (LC3-SWB) shall be included unless it is temporarily unavailable.

  Any speech codecs may be included in this list as long as the respective mandatory codecs are also included.

  Values:

  <u>: All possible Codec IDs. Codec IDs shall be transferred as string representations of decimal numbers. The format of the Codec IDs is 8 bit aliases that are defined in Section 11 (Appendix B: Codec IDs). For a single codec with ID=12 and the mandatory default codecs (1, 2 and 3), the command:

  AT+BAC=1,2,3,12

  is sent.

- **AT+BIND** (Bluetooth HF Indicators Feature)

  Syntax:

  AT+BIND= <a>,<b>,<c>,...,<n> (List HF supported indicators)

  AT+BIND=? (Read AG supported indicators)

  AT+BIND? (Read AG enabled/disabled status of indicators)

  Description:

This command enables the HF to notify the AG which HF to AG indicators are supported. The indicators may be enabled or disabled.

The AT+BIND commands shall not be used unless both AG and HF BRSF bits for the ‘HF Indicators’ feature are set to one.

Values:

<a> ... <n>: 0-65535, entered as decimal unsigned integer values without leading zeros, referencing an HF indicator assigned number. Values are defined on the Bluetooth SIG Assigned Numbers [8] web page. The maximum number of indicator assigned numbers in the request shall be 20.

- **+BIND** (Bluetooth HF Indicators Feature)

Syntax:

+BIND: (<a>,<b>,<c>,...,<n>) (Response to AT+BIND=?)

+BIND: <a>,<state> (Unsolicited or Response to AT+BIND?)

Description:

This response enables the AG to notify the HF which HF indicators are supported and their state, enabled or disabled.

The +BIND responses shall not be used unless both AG and HF BRSF bits for the ‘HF Indicators’ feature are set to one.

When responding to the AT+BIND? command, the AG shall send one or more +BIND responses followed by OK to terminate the list.

Values:

<a> ... <n>: 0-65535, entered as decimal unsigned integer values without leading zeros, referencing an HF indicator assigned number. Values are defined on the Bluetooth SIG Assigned Numbers [8] web page. The maximum amount of indicator assigned numbers in the request shall be 20.

<state>: 0-1, entered as integer values, where

0 = Indicator is disabled, no value changes shall be sent for this indicator

1 = Indicator is enabled, value changes may be sent for this indicator

- **AT+BIEV** (Bluetooth HF Indicators Feature)

Syntax:

AT+BIEV= <assigned number>,<value> (Update value of indicator)

Description:

This command enables the HF to send updated values of the enabled HF indicators to the AG.

The AT+BIEV command shall not be used unless both AG and HF BRSF bits for the ‘HF Indicators’ feature are set to one.

Values:

<assigned number>: 0-65535, entered as a decimal unsigned integer without leading zeros, referencing an HF indicator assigned number. Values are defined on the Bluetooth SIG Assigned Numbers [8] web page.

<value> 0 to 4,294,967,295, entered as a decimal unsigned integer without leading zeros. The meaning of the value depends of the <assigned number> and is defined on the Bluetooth SIG Assigned Numbers [8] web page.

6 RFCOMM

This profile requires compliance to RFCOMM [4]. The following subsections together with the associated sub-clauses defines the requirements with regard to this profile in addition to the requirements as defined in RFCOMM.

6.1 RFCOMM Interoperability Requirements

For the RFCOMM layer, the following requirements apply.

For the Hands-Free Profile, as defined in Section 4.2, both the AG and the HF may initiate RFCOMM connection establishment (see Section 6.1.1). Therefore, for the purposes of reading RFCOMM [4], both the AG and the HF may assume the role of RFCOMM Initiator.

The AG and the HF shall support the RFCOMM Initiator role as defined in Section 9 in [4].

The AG and the HF shall support the RFCOMM Acceptor role, defined in HFP as the device that accepts the establishment of the RFCOMM multiplexer control channel on DLCI0 as defined in Section 5.2.1 in [4].

The AG and the HF shall support the RFCOMM Client role as defined in Section 9 in [4].

The AG and the HF shall support the RFCOMM Server role as defined in Section 9 in [4].

Table 6.1 summarizes the additional requirements for RFCOMM procedure support beyond those defined in RFCOMM [4].

Procedure (see Section 5.2.1 in [4])	RFCOMM Initiator	RFCOMM Acceptor	RFCOMM Client	RFCOMM Server
Initialize RFCOMM session	M	X	X	X
Respond to RFCOMM session initialization	X	M	X	X
Establish DLC	X	X	M	X
Respond to DLC establishment	X	X	X	M

Table 6.1: Additional RFCOMM support requirements

6.1.1 RFCOMM connection establishment

RFCOMM connection establishment means the RFCOMM multiplexer start-up procedure defined in Section 5.2.1 in [4], which describes the establishment of the RFCOMM multiplexer control channel on DLCI0. The device establishing the RFCOMM connection shall first discover the SDP service record as defined in Table 6.3 for the HF and/or in Table 6.5 for the AG to discover the RFCOMM server channel number exposed by the peer device.

The device establishing the RFCOMM connection shall then initiate the L2CAP channel establishment procedure as defined in Section 6.2.1 and shall then perform the RFCOMM multiplexer start-up procedure. When the RFCOMM connection has been established, the device shall proceed to the Service Level Connection Establishment procedure defined in Section 4.2 to establish a Service Level Connection on the RFCOMM server channel discovered during SDP service record discovery.

6.2 L2CAP Interoperability Requirements

For the L2CAP layer, Table 6.2 defines the additional support requirements to those defined in RFCOMM [4].

Procedure	HF requirement	AG requirement
Data Channel Initiator	M	M
Data Channel Acceptor	M	M

Table 6.2: Additional L2CAP support requirements

6.2.1 L2CAP channel establishment procedure

The device establishing the RFCOMM connection as defined in Section 6.1.1 shall send an L2CAP Connection Request and shall set the value of the PSM field to the value for RFCOMM as defined in the Bluetooth Assigned Numbers [8].

6.3 SDP Interoperability Requirements

The following service records are defined for the Hands-Free Profile. There is one service record applicable to the Hands-Free unit and another for the Audio Gateway.

The attribute “SupportedFeatures” states the features supported in each device. This attribute is not encoded as a data element sequence; it is simply a 16-bit unsigned integer. The set of features supported in each case is bit-wise defined in this attribute on a yes/no basis. The mapping between the features and their corresponding bits within the attribute is listed below in Table 6.4 for the HF and in Table 6.6 for the AG. If a device indicates support for a feature, then it shall support that feature in the manner specified by this Profile.

The codes assigned to the mnemonics used in the Value column, as well as the codes assigned to the attribute identifiers (if not specifically mentioned in the AttrID column), are listed in the Bluetooth Assigned Numbers [8] web page.

The values of the “SupportedFeatures” bitmap given in Table 6.4 shall be the same as the values of the Bits 0 to 4 of the AT-command AT+BRSF (see Section 5.3).

Item	Definition	Type	Value	Status	Default
ServiceClassIDList				M	
ServiceClass0		UUID	Hands-Free	M	
ServiceClass1		UUID	Generic Audio	M	
ProtocolDescriptorList				M	
Protocol0		UUID	L2CAP	M	
Protocol1		UUID	RFCOMM	M	
ProtocolSpecificParameter0	Server Channel	Uint8	N=server channel #	M	
BluetoothProfileDescriptorList				M	

Item	Definition	Type	Value	Status	Default
Profile0	Supported Profiles	UUID	Hands-Free	M	Hands-Free
Param0	Profile Version	Uint16	0x0109[6]	M	
ServiceName	Display-able Text name	String	*Service-provider defined*	O	“Hands-Free unit”
SupportedFeatures	Features supported	Uint16	*Device dependent*	M	0x0000

Table 6.3: Service Record for the HF

Bit position (0=LSB)	Feature	Default in HF
0	EC and/or NR function (yes/no, 1 = yes, 0 = no)	0
1	Call waiting or three-way calling(yes/no, 1 = yes, 0 = no)	0
2	CLI presentation capability (yes/no, 1 = yes, 0 = no)	0
3	Voice recognition activation (yes/no, 1= yes, 0 = no)	0
4	Remote audio volume control (yes/no, 1 = yes, 0 = no)	0
5	Wide Band Speech (yes/no, 1 = yes, 0 = no)	0
6	Enhanced Voice Recognition Status (yes/no, 1 = yes, 0 = no)	0
7	Voice Recognition Text (yes/no, 1 = yes, 0 = no)	0
8	Super Wide Band Speech (yes/no, 1 = yes, 0 = no)	0

Table 6.4: “SupportedFeatures” attribute bit mapping for the HF

The "Network" attribute states if the AG has the capability to reject incoming calls[7]. This attribute is not encoded as a data element sequence; it is simply an 8-bit unsigned integer. The information given in the “Network” attribute shall be the same as the information given in Bit 5 of the unsolicited result code +BRSF (see Section 5.3). An attribute value of 0x00 is translated to a bit value of 0; an attribute value of 0x01 is translated to a bit value of 1.

The values of the “SupportedFeatures” bitmap given in Table 6.6 shall be the same as the values of the Bits 0 to 4 of the unsolicited result code +BRSF (see Section 5.3).

[6] Indicating version HFP 1.9

[7] In previous versions of the Hands-Free Profile, the attribute values were called "GSM like" and "others"

Item			Definition	Type	Value	Status	Default
ServiceClassIDList						M	
	ServiceClass0			UUID	AG Hands-Free	M	
	ServiceClass1			UUID	Generic Audio	M	
ProtocolDescriptorList						M	
	Protocol0			UUID	L2CAP	M	
	Protocol1			UUID	RFCOMM	M	
		ProtocolSpecificParameter0	Server Channel	Uint8	N=server channel #	M	
BluetoothProfileDescriptorList						M	
	Profile0		Supported Profiles	UUID	Hands-Free	M	Hands-Free
		Param0	Profile Version	Uint16	0x0109[8]	M	
ServiceName			Display-able Text name	String	*Service-provider defined*	O	"Voice gateway"
Network				Uint8	0x01 – Ability to reject a call 0x00 – No ability to reject a call	M	
SupportedFeatures			Features supported	Uint16	*Device dependent*	M	0x0009

Table 6.5: Service Record for the AG

Bit position (0=LSB)	Feature	Default in AG
0	Three-way calling (yes/no, 1 = yes, 0 = no)	1
1	EC and/or NR function (yes/no, 1 = yes, 0 = no)	0
2	Voice recognition function (yes/no, 1 = yes, 0 = no)	0
3	In-band ring tone capability (yes/no, 1 = yes, 0 = no)	1
4	Attach a phone number for a voice tag (yes/no, 1 = yes, 0 = no)	0
5	Wide Band Speech (yes/no, 1 = yes, 0 = no)	0

[8] Indicating version HFP 1.9

Bit position (0=LSB)	**Feature**	**Default in AG**
6	Enhanced Voice Recognition Status (yes/no, 1 = yes, 0 = no)	0
7	Voice Recognition Text (yes/no, 1 = yes, 0 = no)	0
8	Super Wide Band Speech (yes/no, 1 = yes, 0 = no)	0

Table 6.6: "SupportedFeatures" attribute bit mapping for the AG

6.3.1 Interaction with Hands-Free Profile Rev 0.96 Implementations

HF implementations, which are according to the Hands-Free Profile specification Rev. 0.96, will not send the AT+BRSF command. Likewise, AG implementations, which are according to the Hands-Free Profile specification Rev. 0.96, will not be able to respond to AT+BRSF with the +BRSF unsolicited result code. Instead, they will respond with ERROR.

In order to retrieve the "SupportedFeatures" information from an HF, which does not send AT+BRSF, Service Discovery should be used by the AG implementation. Whenever the "SupportedFeatures" attribute is not present in the HF service record, or if the AG does not perform the Service Discovery procedure, default values as stated in Table 6.4 shall be assumed.

In order to retrieve the "SupportedFeatures" and "Network" information from an AG, which does not send +BRSF, Service Discovery should be used by the HF implementation. Whenever the "SupportedFeatures" attribute is not present in the AG service record, or if the HF does not perform the Service Discovery procedure, default values as stated in Table 6.6 shall be assumed.

6.3.2 Interaction with HFP 0.96, 1.0 and HFP 1.5 implementations

HF implementations that comply with the Hands-Free Profile specification Rev. 0.96,1.0 or 1.5, shall not indicate support for the Codec Negotiation feature and shall neither send the AT+BAC command nor the AT+BCC command to trigger an audio connection establishment by the AG.

AG implementations that comply with the Hands-Free Profile specification Rev. 0.96,1.0 or 1.5, shall not indicate support for the Codec Negotiation feature and shall neither send the +BCS unsolicited response to the HF.

For backward compatibility, HFP Rev. "x.y" implementations shall be able to handle establishment of synchronous connections according to Hands-Free Profile specification Rev. 1.0 or 1.5.

- The HF shall be able to accept establishment of a synchronous connection from an HFP 1.0 or 1.5 AG.
- The AG shall be able to initiate establishment of a synchronous connection to an HFP 1.0 or 1.5 HF.

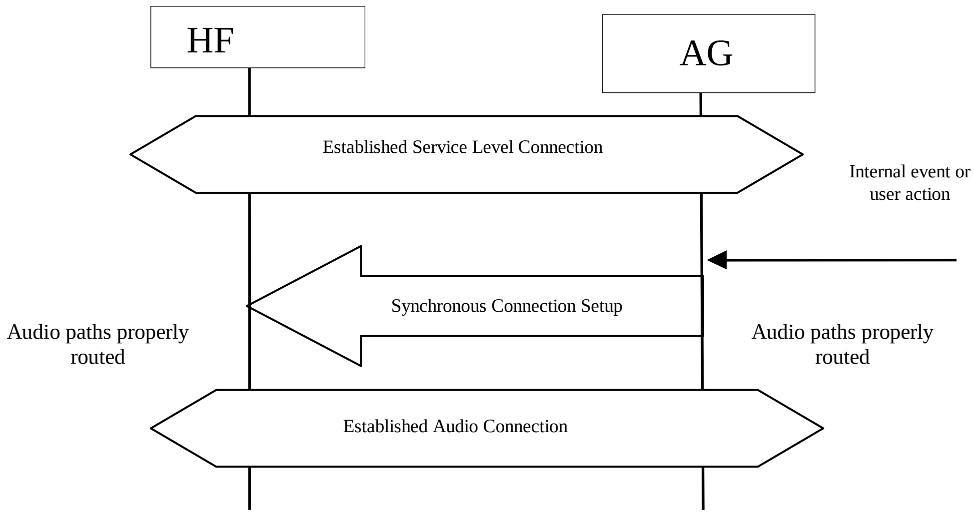

Figure 6.1: Procedure for Establishment of an Audio Connection from AG

The HF shall be able to initiate establishment of a synchronous connection to an HFP 1.0 or 1.5 AG.

The AG shall be able to accept establishment of a synchronous connection from an HFP 1.0 or 1.5 HF.

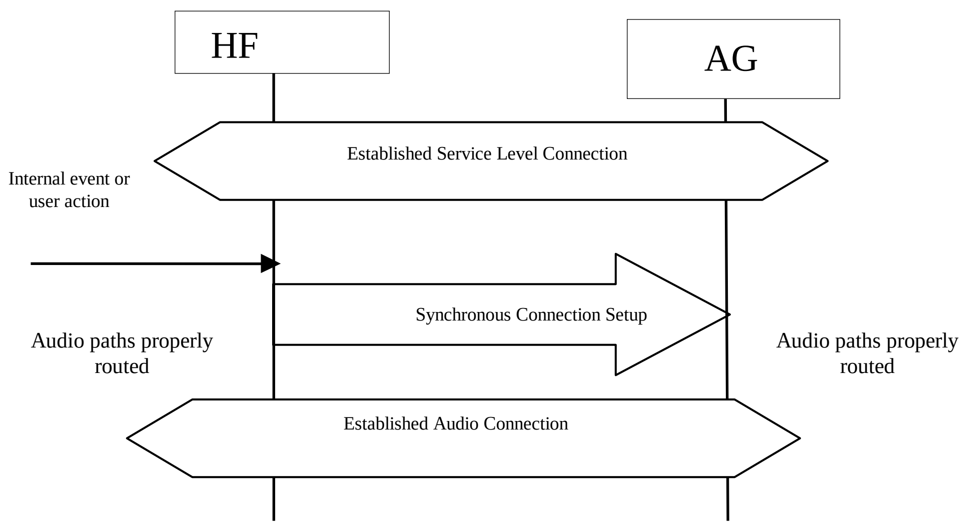

Figure 6.2: Procedure for Establishment of an Audio Connection from HF

6.4 Link Manager (LM) Interoperability Requirements

This profile requires support for encryption as defined in the Link Manager Protocol specification in Volume 2, Part C, Section 3.5 in [1].

Additionally this profile mandates that both the AG and HF devices shall support synchronous logical transports, subject to the requirements in Section 6.6.

6.5 Link Control (LC) Interoperability Requirements

Table 6.7 shows the Link Controller (See Volume 2, Part B, Section 8 in [1]) interoperability requirements for HFP.

	Capability	Support in AG	Support in HF
1.	Inquiry		O
2.	Inquiry scan	O	
7	Voice CODEC		
C	CVSD	M	M

Table 6.7: Link Controller requirements

6.5.1 Class of Device

A device implementing the HF role of HFP shall set the "Audio" bit in the Service Class field. Optionally, if the HF intends to be discovered as a "Hands-Free", it may use the following values in the Class of Device field:

1. Indicate "Audio" as Major Device class.
2. Indicate “Hands-Free” as the Minor Device class.

An inquiring AG may use this information to filter the inquiry responses.

6.6 Baseband Interoperability Requirements

The eSCO link and the air mode “Transparent Data” shall be supported if Wide Band Speech or Super Wide Band Speech is supported and are optional otherwise.

Erroneous data delivery may be used for Wide Band Speech only if core version 2.1 or higher is supported.

Item	Capability	Status
1.1.3/3	Support of eSCO link	M
1.1.7/4	Transparent data	C1
1.1.8/2	Erroneous data delivery	C2

C1: Mandatory if Wide Band Speech or Super Wide Band Speech is supported, or optional otherwise.

C2: Optional if core version 2.1 or higher is supported, or excluded otherwise

Table 6.8: Baseband Requirements

6.7 Codec Interoperability Requirements

Table 6.9 shows supported *Mandatory* and *Optional* codecs in this profile.

Codec Type	Support	Ref
CVSD	M	6.7.3
mSBC	C1	6.7.4
LC3-SWB	C2	6.7.6

C1: Mandatory if Wide Band Speech is supported or excluded otherwise.

C2: Mandatory if Super Wide Band Speech is supported or excluded otherwise

Table 6.9: Supported codecs

6.7.1 Synchronous Connection Interoperability Requirements

6.7.1.1 Selection of Synchronous Transport

The device starting the request for a Synchronous Connection is known as the Initiator and the device receiving the request is known as the Responder. The Responder may accept or reject a request for a logical transport

To select the type of synchronous transport (eSCO or SCO) to use, devices shall adhere to the following logic:

- If eSCO is supported by the responder, the synchronous connection shall first be attempted on an eSCO logical transport. See section 6.7.1.2.
- If eSCO is unavailable for use (e.g., not supported by the Responder or link establishment fails), and SCO is not currently forbidden because a BR/EDR secure connection is being used, the Initiator shall open a SCO logical connection. See section 6.7.1.3.

6.7.1.2 Negotiation of eSCO Configuration Parameters

In this section, HCI level parameters are given as a reference. On systems not incorporating HCI, equivalent values for LMP level eSCO connection parameters T_{eSCO}, W_{eSCO} and packet length that correspond to those HCI parameters shall be used. The following requirements apply to the support and use of eSCO logical transports and are based on parameter sets S1-S4 and T1-T2:

- The Initiator's request for an eSCO transport may involve any configuration parameters matching the bidirectional throughput requirements of the voice codec (see Section 6.5). If an HCI is supported on this device, the request for setting up a synchronous connection may include single or multiple packet types masked within the same request.
- The Responder may choose to accept or reject the request from the Initiator. It may reject the request for an eSCO transport, or may accept it with parameters that do not match the requested parameters. In this case, the Initiator may retry the Synchronous Connection setup with different configuration parameters.
- It is recommended to use the S4 or T2 settings when BR/EDR Secure Connections is in use, when using complicated topologies, or when a device wants to improve coexistence with other wireless technologies.
- If the request for S4, S3 or T2 has failed and the Initiator needs to employ a time-division based solution to MWS Coexistence, the Initiator should make its best efforts to reconfigure its MWS radio so that it no longer needs to employ a time-division based solution to MWS Coexistence. For example, an AG might have the opportunity to change the cellular frequency band it is using in order to allow the MWS Coexistence features to be turned off for the duration of the synchronous connection.
- A device shall accept to use the S4 or T2 settings when requested by the remote device. Devices supporting only 1.7.1 or earlier versions of this profile might refuse the S4 setting, in which case the initiating device should request the S3 setting.

- If a request for an eSCO logical transport fails, the Initiator shall not abandon the setup of an eSCO transport without having requested eSCO using the “safe settings” specified in the following table:

	CVSD coding	mSBC coding	LC3-SWB coding
If BR/EDR Secure Connections is not in use on the connection	**S1**	**T1**	**T1**
If BR/EDR Secure Connections is in use on the connection	**S4**	**T2**	**T2**

Table 6.10: Mandatory Safe Settings

- Only for HCI-based devices: if the Responder does not reject the request for an eSCO transport, the response shall always include the parameters corresponding to the “safe settings” specified in Table 6.10 when accepting a request. If the Initiator fails to establish an eSCO transport with the S1 settings, the Initiator shall request the setup of a SCO transport, if not currently forbidden because a BR/EDR secure connection is being used.
- If Wide Band Speech or Super Wide Band Speech is supported, and the Initiator fails to establish an eSCO transport with the T1 settings, or the Initiator fails to establish an eSCO transport with the T2 settings if Secure Connections is in use, the Initiator shall attempt to set up a Codec Connection using the narrow band CVSD codec over eSCO (or SCO transport if not currently forbidden because a BR/EDR secure connection is being used).

6.7.1.3 Negotiation of SCO Configuration Parameters

Requirements related to the use of SCO links, under the conditions when the use of a SCO logical transport is permitted, are covered by the parameter sets D0-D1.

6.7.2 Synchronization Header for Transparent Data

Two synchronization headers have been defined; the first one only contains a sequence number. The second one has both a synchronization word and a sequence number.

H1: Header with sequence number

Figure 6.3 shows the layout of the header H1 that only contains the sequence number. The one octet header shall contain a 4 bit sequence number (SN0,…, SN3). The sequence number is protected by a simple repetition code (all bits are duplicated). Hence, each sequential pair of bits in the sequence number shall be always 00 or 11.

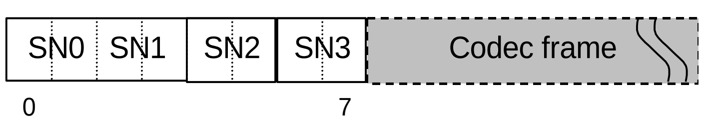

Figure 6.3: One octet frame synchronization H1 header

When using the Erroneous data delivery feature, the sequence number in the H1 header shall always be different from 0. The reason is that the Erroneous data delivery feature sets 0 for payload data octets corresponding to missing (e)SCO packets.

H2: Header with synchronization word and sequence number

Figure 6.4 shows the layout of the header H2 that contains both the synchronization word and the sequence number. The two octet header shall contain a 12 bit synchronization word and a 2 bit sequence number (SN0, SN1). The latter is protected by a simple repetition code (both bits are duplicated). Hence, each pair of bits in the sequence number shall be always 00 or 11.

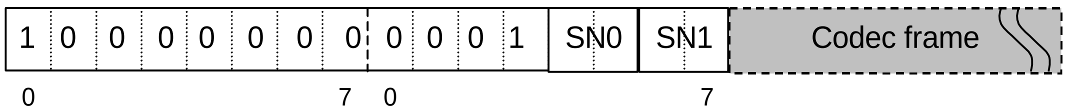

Figure 6.4: Two octet frame synchronization H2 header

It should be noted that when the transparent data transport is used, any alignment of the coded audio stream frames (codec frames and synchronization word) with eSCO packet boundaries is left up to the implementation. The use of the synchronization header enables unaligned codec audio frames to be recovered by the receiving side. To successfully render audio, the receiving side shall not assume the location of the synchronization header within the eSCO packets. The receiving side shall be capable of rendering received audio.

6.7.3 CVSD coding

Table 6.11 defines SCO configuration parameter sets D0 and D1 for usage of CVSD coding.

SCO parameter set	**D0**	**D1**
Packet type	HV1	HV3
Transmit/Receive Bandwidth	8000	8000
Voice_Setting (air coding)	CVSD	CVSD
Max_Latency	0x0004 (4 ms)	0x0005 (5 ms)
Retransmission_Effort	0x00 or 0xFF	0x00 or 0xFF

Table 6.11: SCO synchronous connections (HCI Reference parameters)

Table 6.12 defines eSCO configuration parameter sets S1, S2, S3 and S4 for usage of CVSD coding.

eSCO parameter set	**S1**	**S2**	**S3**	**S4**
Packet type	EV3	2-EV3	2-EV3	2-EV3
Transmit/Receive Bandwidth	8000	8000	8000	8000
Voice_Setting (air coding)	CVSD	CVSD	CVSD	CVSD
Max_Latency	0x0007 (7 ms)	0x0007 (7 ms)	0x000A (10ms)	0x000C (12ms)
Retransmission_Effort	0x01	0x01	0x01	0x02

Table 6.12: eSCO synchronous connections (HCI Reference parameters)

6.7.4 mSBC coding

Support for a modified version of the SBC codec (hereafter called mSBC) is mandatory if Wide Band Speech is supported in this profile. The original SBC codec is specified in A2DP [7]. The modifications to SBC that constitute mSBC are specified in Appendix A of this profile.

The various mandatory and optional settings for this codec are specified in the tables below.

Support for the following mSBC configuration is mandatory if Wide Band Speech is supported.

Parameter	Value
Channel mode	Mono
Sampling rate	16 kHz
Allocation method	Loudness
Subbands	8
Blocklength	15
Bitpool	26

Table 6.13: mSBC mandatory parameter set

An mSBC frame shall have the synchronization header H2 before every frame. The H2 header allows the receiver to know if one or more radio link packets are dropped.

Wide Band Speech, when using the mSBC codec, shall only use the eSCO transport. The eSCO T1 setting below shall be supported.

eSCO parameter set	T1	T2
Packet type	EV3	2-EV3
Transmit/Receive Bandwidth	8000	8000
Voice_Setting (air coding)	Transparent Data	Transparent Data
Max_Latency	8 ms	13 ms
Retransmission_Effort	0x02	0x02

Table 6.14: eSCO parameters for a Wide Band Speech connection using the mSBC codec or a Super Wide Band Speech connection using the LC3-SWB codec.

6.7.5 Codec vs Link Parameter Negotiation

The Codec Negotiation procedure defined in this profile determines which codec to use when establishing an audio connection. This procedure is not used to negotiate the link parameters used with the selected codec. The link parameters are negotiated by the link managers using the link manager protocol.

The recommended link parameters are:

- For CVSD: S4 is preferred over S3, S3 is preferred over S2, S2 is preferred over S1, S1 is preferred over D1 and D1 is preferred over D0. The decision to use S4 may also be based on implementation-specific coexistence use cases.
- For mSBC: T2 is preferred over T1.

- For LC3-SWB: T2 is preferred over T1.

6.7.6 LC3-SWB coding

Support for the LC3-SWB codec is mandatory if Super Wide Band Speech is supported in this profile. The codec is specified in the LC3 Specification [10]. The AG and the HF shall implement both the encoder and decoder of the LC3-SWB codec.

The settings for LC3-SWB are specified in Table 6.14 and Table 6.15.

Support for the configuration shown in Table 6.15 is mandatory if Super Wide Band Speech is supported.

Parameter	Value
Channel mode	Mono
Sampling rate	32 kHz
Frame duration	7.5 ms
Byte count per codec frame (excluding H2 header)	58
Resulting bit rate (excluding H2 header)	61867 bits/s

Table 6.15: LC3-SWB mandatory parameter set

See Section 3.2.5 in [10] for the relation between the values of bit rate and the byte count.

An LC3-SWB frame shall have the synchronization header H2 before every frame. The 2-byte H2 header enables the receiver to determine if one or more radio link packets are dropped.

Super Wide Band Speech, when using LC3-SWB, shall use only the eSCO transport. The eSCO T1 setting shown in Table 6.14 shall be supported.

6.8 Speech Quality Recommendations

The introduction of Wide Band Speech and Super Wide Band Speech to the HFP profile allows for increased end-customer satisfaction in terms of perceived speech quality. Super Wide Band Speech corresponds to the maximum audio bandwidth provided in 5G voice services and provided by many over-the-top (OTT) voice services. The available codec with the highest audio quality should be used. For example, Super Wide Band Speech is preferred over Wide Band Speech, and Wide Band Speech is preferred over narrow band speech connections.

6.8.1 Packet Loss Concealment (PLC)

It is strongly recommended that some form of PLC should be implemented on the receiving ends of compressed audio (such as mSBC and LC3-SWB) to reduce synthesized audio artifacts during degraded channel conditions.

For mSBC, the example PLC algorithm provided in Appendix C may be used. The speech quality of this example PLC under typical packet loss conditions is considered satisfactory. If implementations choose to modify or implement an alternate PLC scheme, the performance of any such alternate PLC should meet or exceed the performance of the example PLC provided in Appendix C.

The LC3 specification [10] also provides an example PLC algorithm that may be used with Super Wide Band Speech (see Appendix B in [10]).

6.8.2 Signal Levels

Full swing at the 16 bit linear PCM interface to the codecs in this profile specification is defined to be 3 dBm0. This definition applies to devices that support narrow band, Wide Band Speech, and Super Wide Band Speech.

Further details on recommended audio levels are specified in Appendix D, Section 13.1.

Recommended specifications for send and receive frequency masks are provided in Appendix D, Section 13.2.

7 Generic Access Profile

This section defines the support requirements for the capabilities as defined in the "Generic Access Profile" of the Core Specification.

7.1 Modes

Table 7.1 shows the support status for GAP Modes in this profile.

Procedure	**Support in HF**
General discoverable mode	M
Procedure	**Support in AG**
Pairable mode	M

Table 7.1: Modes

7.2 Security Aspects

Baseband authentication and encryption is optional for both the Hands-Free unit and the Audio Gateway. If both devices support authentication and encryption, the application on either device may require its use. However, if Secure Connections is supported by both devices, its use is mandated by the Bluetooth Core 4.1 Specification (or later).

A Hands-Free unit may be able to use the services of the Audio Gateway without the creation of a secure connection. It is implementation specific whether the Hands-Free unit provides or supports security enforcement for the user.

Whenever baseband authentication or encryption is used, the two devices shall create a secure connection using the GAP authentication procedure as described in Generic Access Profile (Volume 3, Part C, Section 5.1 in [1]). This procedure may include entering a Bluetooth PIN code or passkey and creation of proper link keys. If Simple Secure Pairing is not in use and when the UI of the Hands-Free unit is limited, a fixed Bluetooth PIN code may be used during the GAP authentication procedure.

If Secure Connections is used, the Authenticated Payload Timeout should be less than or equal to 10s. The disconnection behavior following a timeout is implementation specific.

7.3 Idle Mode Procedures

Table 7.2 shows the support status for Idle mode procedures within this profile.

Procedure	**Support in AG**
Initiation of general inquiry	M
Initiation of general bonding	O
Initiation of dedicated bonding	O

Table 7.2: Idle mode procedures

8 References

[1] Bluetooth Core Specification, Version 4.2 or later

[2] 3GPP 27.007 v6.8.0 now supersedes and replaces ETS 300 916, "Digital cellular telecommunications system (Phase 2+); AT command set for GSM Mobile Equipment (ME) (GSM 07.07 version 7.5.0)," http://www.3gpp.org/ftp/Specs/html-info/27007.htm

[3] Headset Profile Specification, Version 1.1 or later

[4] RFCOMM with TS 07.10 Specification, Version 1.2 or later

[5] ITU-T50, Terminal Equipment and Protocols for telematic services: International Reference Alphabet (IRA) (Formerly International Alphabet No. 5 IA5). Information technology – 7-Bit coded character set for information interchange

[6] Digital cellular telecommunication system (Phase 2+); Mobile radio interface layer 3 specification, (GSM 04.08 version 6.11.0)

[7] Advanced Audio Distribution Profile Specification, Version 1.2 or later

[8] Bluetooth Assigned Numbers, https://www.bluetooth.com/specifications/assigned-numbers

[9] D. Goodman, et al., "Waveform Substitution Techniques for Recovering Missing Speech Segments in Packet Voice Communications", IEEE Transaction on Acoustics, Speech and Signal Processing, December 1986, pp. 1440 - 1448.

[10] Low Complexity Communication Codec Specification, Version 1.0

[11] Appropriate Language Mapping Tables, https://www.bluetooth.com/language-mapping/Appropriate-Language-Mapping-Table

9 List of Acronyms and Abbreviations

Abbreviation or Acronym	Meaning
AG	Audio Gateway
AT	Attention
BTR	Bluetooth Reference Point
BVRA	Bluetooth Voice Recognition Activation
CLI	Calling Line Identification
CODEC	COder DECoder
CVSD	Continuously Variable Slope Delta modulation
DTMF	Dual Tone Multi-Frequency
EC	Echo Cancellation
EDR	Enhanced Data Rate
eSCO	Extended Synchronous Connection Oriented
GAP	Generic Access Profile
GSM	Global System for Mobile communication
HF	Hands-Free unit
L2CAP	Logical Link Control and Adaptation Protocol
LC3	Low Complexity Communication Codec
LMP	Link Manager Protocol
mSBC	Modified Sub Band Codec
NR	Noise Reduction
OSI	Open System Interconnection
OTT	Over-the-top
PCM	Pulse Code Modulation
PLC	Packet Loss Concealment
PIN	Personal Identification Number
POI	Point Of Interconnection
RFCOMM	Serial port transport protocol over L2CAP
SBC	Sub Band Codec
SCO	Synchronous Connection Oriented
SDP	Service Discovery Protocol
SWB	Super Wide Band

Abbreviation or Acronym	Meaning
UI	User Interface
UUID	Universally Unique Identifier
VR	Voice Recognition

10 Appendix A: Technical Specification of mSBC

10.1 Introduction

This appendix describes the changes to the SBC codec defined in A2DP [7] standard which comprises the Modified SBC codec. The changes to the A2DP SBC are limited to the frame header syntax and semantics. All other parts of the SBC definition remain un-modified.

10.1.1 Mnemonics

The following mnemonics are defined to describe the different data types used in the coded bit-stream.

Mnemonic	Description
Char8	Character of 8 bits
UiMsbf	Unsigned integer, Most significant bit first
SiMsbf	Signed integer, Most significant bit first
BsMsbf	Bit-stream, Most significant bit first
PCM	Pulse Code Modulation
Na	Not available

Table 10.1: Mnemonics

10.2 Syntax

Syntax	No. of bits	Mnemonic

audio_frame()
{
    frame_header()
    scale_factors()
    audio_samples()
    padding()
}		

Table 10.2: Syntax of mSBC speech frame

Note: The syntax of the mSBC speech frame is not changed from the original SBC definition in A2DP [7].

Syntax	No. of bits	Mnemonic
frame_header()		
{		
Syncword	8	BsMsbf
Reserved for future use	16	UiMsbf
crc_check	8	UiMsbf
}		

Table 10.3: Syntax of mSBC frame header

10.3 Semantics

10.3.1 Frame_header

syncword -- The 8 bit string %10101101 or $AD.

The syncword is different from that specified for the SBC in A2DP specifications so that implementations may use this is an additional indication that the encoded stream is meant to contain Wide Band Speech.

Codec ID – The following table represents the values of the various mSBC parameters that correspond to the values specified in the Codec ID field.

8 bit Codec ID	Mandatory Or Optional	mSBC Parameters					
		Channel Mode	Sampling rate	Allocation method	Subbands	Block length	Bitpool
0x02	M	Mono	16 kHz	loudness	8	15	26

Table 10.4: Mapping of Codec Ids to mSBC parameters

The interpretation of the mSBC parameters as specified in Table 10.4 is identical to that specified in SBC definition in A2DP [7].

10.3.2 Padding

The all zero 8 bit string %00000000 or $00 shall be used for padding.

11 Appendix B: Codec IDs

The following table specifies the mapping of the codecs to their respective Codec Ids.

Codec Type	Codec ID	Ref.
CVSD	0x01	6.7.3
mSBC	0x02	6.7.4
LC3-SWB	0x03	6.7.6
Other optional codecs may be added here	Reserved	Reserved

Table 11.1: Mapping of codec types to Codec IDs

12 Appendix C: Example PLC Implementation

This appendix describes a packet loss concealment (PLC) method based on the paper by D. Goodman, et al. [9] and adapted for use with mSBC.

12.1 Baseline Packet Loss Concealment

A baseline packet loss concealment algorithm based on the techniques presented in [1] is discussed. This baseline PLC is a generic PCM-based system and makes no assumptions on mSBC. Its integration into mSBC to form a reference PLC for mSBC is described in a later section.

The following subsections describe the techniques used in the baseline PLC algorithm.

12.1.1 Waveform Substitution Based On Pattern Matching

The technique in Section IV “Waveform Substitution Based on Pattern Matching” of [1] is used to construct an estimate of the waveform in missing packets. This algorithm searches a history buffer of previously received samples to find a packet-worth of samples to replace the missing packet. The selection is based on constructing a template of M samples immediately preceding the missing packet and scanning a search window of N samples in the history buffer to find the M samples that best match the template. It then uses as a replacement packet the L samples that follow the best match. The pattern matching criterion used is the cross-correlation similar to that in equation (4) of the paper except the denominator contains a square root as in the standard definition of cross-correlation. The implementation follows the recommendations of the paper and uses a template of 4ms (M=64) and a search window of 16ms (N=256). A frame size of 7.5ms (L=120) was used corresponding to the frame size of the default mSBC configuration.

12.1.2 Overlap-Add

As described in Section III of the paper, in order to reduce the audible distortion caused by discontinuities at the boundaries between correct packets and substitution packets, it is important to “merge” the two waveforms by way of an overlap-add procedure. The example PLC follows the recommendation of the paper and uses a raised-cosine weighting with a merge duration of 1ms (16 samples). The paper adds 1ms of delay in order to merge at the boundary between the last correct packet and the substitution packet. However, the example PLC avoids this added delay by using the partial signal contained in the mSBC decoder memory. This will be described in more detail in Section 12.2.1. At the boundary between the first correct packet, after packet loss, and the substitution packet, the paper extends the substitution waveform 1ms into the correct packet in order to perform the merge. The example PLC follows a similar method except that the substitution waveform is further extended to allow for the mSBC decoder signal to re-\converge after packet loss. This is also described later.

12.1.3 Amplitude Matching

Section B of the paper describes a procedure to adjust the amplitude of the substitution packet to match that of the preceding packet. The example PLC implements this approach with the mean-absolute-value measure of packet amplitude.

12.2 Integration of PLC with mSBC

The PLC described in the paper was designed for PCM without any considerations for integration with a codec such as mSBC. The main issue with mSBC as far as integration of a PLC is the response of the sub-band filters in the mSBC decoder. The integration is described in the next sub-sections.

12.2.1 Merging in the First Substitution Frame – Avoiding Delay

As previously mentioned, the paper performs a merge between the last correct packet and the substitution packet by introducing a 1ms delay. This delay is avoided in the example PLC by taking advantage of the signal embedded in the mSBC decode sub-band filter memory. In the first substitution frame, the mSBC decoder is fed a bit-stream corresponding to an all-zero input signal. The output of decoder is then used to merge with the substitution waveform using a 1ms raised-cosine weighting as described in the paper. This procedure also has the advantage of setting the decoder memory to zero.

12.2.2 Reconvergence of the mSBC Decoder in the First Correct Packet After Packet Loss

The delay caused by the sub-band filters in the mSBC decoder means that when the first correct packet after packet loss is decoded, there will be a period of time before the output reconverges to the desired decoder output. Since the decoder memory is essentially reset to zero by the procedure described in Section 12.2.1, the decoder output will ramp up from zero as it reconverges. The substitution waveform is extended into the first correct packet after packet loss by an amount equal to this reconvergence time and used as output. Once the decoder output has sufficiently reconverged, the substitution waveform is merged with the decoder output using a 1 ms raised-cosine weighting. The reconvergence time used was experimentally tuned to 36 samples (2.25ms).

12.3 API Description

This section describes the API for the example PLC for mSBC.

12.3.1 Memory Allocation

The calling application allocates memory for the PLC memory structure.

The structure is defined in the file sbcplc.h and is shown below:

```
/* PLC State Information */
struct PLC_State
{
   short hist[LHIST+FS+SBCRT+OLAL];
   short bestlag;
   int   nbf;
};
```

The main test program can allocate the space by simply declaring a variable as below:

```
struct PLC_State plc_state;
```

12.3.2 Initialization

The calling application initializes the PLC memory that was allocated as described in Section 12.3.1. The initialization function is called once before processing begins. The initialization function is:

```
void InitPLC(struct PLC_State *plc_state)
```

and is found in sbcplc.c.

The calling program should include `sbcplc.h` containing the header:

```
#include "sbcplc.h"
```

and should call the initialization function:

```
InitPLC(&plc_state);
```

12.3.3 Good Frame Processing

If a good frame is received, the frame should be decoded by mSBC. The output PCM samples from the mSBC decoder are then passed to the function:

```
void PLC_good_frame(struct PLC_State *plc_state, short *in, short *out)
```

where:

`struct PLC_State *plc_state`	the PLC memory block
`short *in`	Pointer to the array of samples from the mSBC decoder containing the PCM samples of the good decoded frame.
`short *out`	Pointer to the array of samples processed by `PLC_good_frame()`.

The calling application owns the memory pointed to by **in* and **out.*

- The size of these arrays should be no smaller than the frame size.
- The samples pointed to by `*out` are the final output samples.

12.3.4 Bad Frame Processing

If a frame is lost, the calling application does the following:

Call the mSBC decoder with channel indices representing an all-zero input PCM signal. This is further described in Section 12.3.5

Call the mSBC PLC function

```
void PLC_bad_frame(struct PLC_State  *plc_state, short *ZIRbuf, short
*out)
```

where:

`struct PLC_State *plc_state`	the PLC memory block.
`short *ZIRbuf`	the zero input response of the mSBC decoder from step 1) of length equal to the frame size.
`short *out`	pointer to the buffer where the PLC will write the concealment samples for the current lost frame.

12.3.5 SBC Decoder Zero-Input Response

At time of receiving the first bad frame, the mSBC decoder filter banks contain data from previous good frames, called signal memory. This signal memory can be extracted by calling the mSBC decoder with a bit-stream that represents an all-zero PCM signal. The resulting signal is termed the Zero-Input Response (ZIR) of the mSBC decoder. The example PLC requires this signal as part of its input.

The indices representing the all-zero signal can be obtained offline and stored. The indices are then used as input to the decoder every time a frame is lost.

To obtain the indices, use an all-zero PCM signal as input to the encoder, and capture the indices. Store these indices and use them as described above.

The zero input bit stream for the default mSBC configuration is pre-computed and provided below:

```
#if FS==120  /* Frame Size of 120 samples

#define FSIDX        57              /* Frame Size Indexes*/

#define NROFBLK      15              /* NumbeR OF BLocKs*/

#define CHMODE       0               /* CHannel MODE   0=mono, 1=dual,
2=stereo, 3=joint   */

#define ALLOCMETHOD  0               /* bit ALLOCation METHOD  0=loudness,
1=SNR           */

#define NROFSB       8               /* NumbeR OF SubBands */

#define BITPOOL      26              /* BITPOOL size */

#endif

short indices0[] = {0xad, 0x0, 0x0, 0xc5, 0x0, 0x0, 0x0, 0x0, 0x77, 0x6d,
0xb6, 0xdd, 0xdb, 0x6d, 0xb7, 0x76, 0xdb, 0x6d, 0xdd, 0xb6, 0xdb, 0x77, 0x6d,
0xb6, 0xdd, 0xdb, 0x6d, 0xb7, 0x76, 0xdb, 0x6d, 0xdd, 0xb6, 0xdb, 0x77, 0x6d,
0xb6, 0xdd, 0xdb, 0x6d, 0xb7, 0x76, 0xdb, 0x6d, 0xdd, 0xb6, 0xdb, 0x77, 0x6d,
0xb6, 0xdd, 0xdb, 0x6d, 0xb7, 0x76, 0xdb, 0x6c};
```

12.3.6 Bad Frame Calling Example

In the case of a bad frame, using the above example data, the calling application would do the following:

```
SBCdecode( indices0, ZIRbuf);

PLC_bad_frame(plc_state, ZIRbuf, outbuf);
```

The application would then use the contents of *outbuf* for playback.

12.4 Source Code (ANSI C)

12.4.1 Source code for file – sbcplc.h

```
/********************************************************

SBC Example PLC ANSI-C Source Code

File: sbcplc.h

********************************************************/

#ifndef SBCPLC_H

#define SBCPLC_H
```

```
#define FS        120 /* Frame Size */
#define N         256 /* 16ms - Window Length for pattern matching */
#define M         64 /* 4ms - Template for matching */
#define LHIST     (N+FS-1) /* Length of history buffer required */
#define SBCRT     36 /* SBC Reconvergence Time (samples) */
#define OLAL      16 /* OverLap-Add Length (samples) */

/* PLC State Information */
struct PLC_State
{
   short hist[LHIST+FS+SBCRT+OLAL];
   short bestlag;
   int   nbf;
};

/* Prototypes */
void InitPLC(struct PLC_State *plc_state);
void PLC_bad_frame(struct PLC_State  *plc_state, short *ZIRbuf, short *out);
void PLC_good_frame(struct PLC_State *plc_state, short *in, short *out);

#endif /* SBCPLC_H */
```

12.4.2 Source code for the file – sbcplc.c

```
/*************************************************************
   SBC Example PLC ANSI-C Source Code
   File: sbcplc.c
*************************************************************/
#include <math.h>
//#include "sbc.h"
#include "sbcplc.h"
/* Local Function Prototypes */
float CrossCorrelation(short *x, short *y);
int PatternMatch(short *y);
```

```
float AmplitudeMatch(short *y, short bestmatch);

/* Raised COSine table for OLA */
float rcos[OLAL] = {0.99148655f,0.96623611f,0.92510857f,0.86950446f,
   0.80131732f,0.72286918f,0.63683150f,0.54613418f,
   0.45386582f,0.36316850f,0.27713082f,0.19868268f,
   0.13049554f,0.07489143f,0.03376389f,0.00851345f};
+
/***************************************************************************
 * Function: InitPLC() *
 * Purpose: Perform PLC initialization of memory vectors. *
 * Inputs:  *plc_state - pointer to PLC state memory *
 * Outputs: *plc_state - initialized memory. *
 * Date: 03-18-2009

***************************************************************************/
void InitPLC(struct PLC_State *plc_state)
{
int i;
plc_state->nbf=0;
plc_state->bestlag=0;
for (i=0;i<LHIST+SBCRT;i++)
   plc_state->hist[i] = 0;
}
/***********************************************************
 * Function: PLC_bad_frame()
 *
 * Purpose: Perform bad frame processing.
 *
 * Inputs:  *plc_state - pointer to PLC state memory
 *     *ZIRbuf    - pointer to the ZIR response of the SBC decoder
 *
 * Outputs: *out - pointer to the output samples
*
```

```
 * Date: 03-18-2009
 ************************************************************/
void PLC_bad_frame(struct PLC_State  *plc_state, short *ZIRbuf, short *out)
{
   int   i;
   float val;
   float sf;

   plc_state->nbf++;

   sf=1.0f;
   i=0;
   if (plc_state->nbf==1)
   {
  /* Perform pattern matching to find where to replicate */
  plc_state->bestlag = PatternMatch(plc_state->hist);
  plc_state->bestlag += M; /* the replication begins after the template match
*/

/* Compute Scale Factor to Match Amplitude of Substitution Packet to that of
Preceding Packet */
      sf = AmplitudeMatch(plc_state->hist, plc_state->bestlag);

      for (i=0;i<OLAL;i++)
      {
         val = ZIRbuf[i]*rcos[i] + sf*plc_state->hist[plc_state-
          >bestlag+i]*rcos[OLAL-i-1];
         if (val >  32767.0) val= 32767.0;
         if (val < -32768.0) val=-32768.0;
         plc_state->hist[LHIST+i] = (short)val;
      }
      for (;i<FS;i++)
      {
         val = sf*plc_state->hist[plc_state->bestlag+i];
```

```
            if (val >  32767.0) val= 32767.0;
            if (val < -32768.0) val=-32768.0;
            plc_state->hist[LHIST+i] = (short)val;
         }
         for (;i<FS+OLAL;i++)
         {
            val = sf*plc_state->hist[plc_state->bestlag+i]*rcos[i-FS]+plc_state-
>hist[plc_state->bestlag+i]*rcos[OLAL-1-i+FS];
            if (val >  32767.0) val= 32767.0;
            if (val < -32768.0) val=-32768.0;
            plc_state->hist[LHIST+i] = (short)val;
         }
         for (;i<FS+SBCRT+OLAL;i++)
            plc_state->hist[LHIST+i] = plc_state->hist[plc_state->bestlag+i];
      }
      else
      {
         for (;i<FS;i++)
            plc_state->hist[LHIST+i] = plc_state->hist[plc_state->bestlag+i];
         for (;i<FS+SBCRT+OLAL;i++)
            plc_state->hist[LHIST+i] = plc_state->hist[plc_state->bestlag+i];
      }


      for (i=0;i<FS;i++)
         out[i] = plc_state->hist[LHIST+i];

      /* shift the history buffer */
      for (i=0;i<LHIST+SBCRT+OLAL;i++)
         plc_state->hist[i] = plc_state->hist[i+FS];
}
```

```
/*****************************************************************************
 * Function: PLC_good_frame()
 *
 * Purpose: Perform good frame processing. Most of the time, this function
 * just updates history buffers and passes the input to the output,
 * but in the first good frame after frame loss, it must conceal the
 * received signal as it reconverges with the true output.
 *
 * Inputs:  *plc_state - pointer to PLC state memory
 *      *in     - pointer to the input vector
 *
 * Outputs: *out - pointer to the output samples
 * Date: 03-18-2009
 
*****************************************************************************/
void PLC_good_frame(struct PLC_State *plc_state, short *in, short *out)
{
   int i;

   i=0;
   if (plc_state->nbf>0)
   {
      for (i=0;i<SBCRT;i++)
         out[i] = plc_state->hist[LHIST+i];
      for (;i<SBCRT+OLAL;i++)
         out[i] = (short)(plc_state->hist[LHIST+i]*rcos[i-SBCRT] +
in[i]*rcos[OLAL-1-i+SBCRT]);
   }
   for (;i<FS;i++)
      out[i] = in[i];

   /*Copy the output to the history buffer */
   for (i=0;i<FS;i++)
      plc_state->hist[LHIST+i] = out[i];
```

```
   /* shift the history buffer */
   for (i=0;i<LHIST;i++)
      plc_state->hist[i] = plc_state->hist[i+FS];

   plc_state->nbf=0;
}
/*****************************************************************************
 * Function: CrossCorrelation()
 *
 * Purpose: Compute the cross correlation according to Eq. (4) of Goodman
 *     paper, except that the true correlation is used. His formula
 *     seems to be incorrect.
 *
 * Inputs:  *x    - pointer to x input vector
 *     *y    - pointer to y input vector
 *
 * Outputs: Cn    - return value containing the cross-correlation of x and y
*
 * Date: 03-18-2009

*****************************************************************************/
float CrossCorrelation(short *x, short *y)
{
   int   m;
   float num;
   float den;
   float Cn;
   float x2, y2;

   num=0;
   den=0;

   x2=0.0;
```

```
    y2=0.0;
    for (m=0;m<M;m++)
    {
        num+=((float)x[m])*y[m];
        x2+=((float)x[m])*x[m];
        y2+=((float)y[m])*y[m];
    }
    den = (float)sqrt(x2*y2);

    Cn = num/den;
    return(Cn);
}


/***************************************************************************
* Function: PatternMatch()
*
* Purpose: Perform pattern matching to find the match of template with the
* history buffer according to Section B of Goodman paper.
*
* Inputs:  *y : pointer to history buffer
*
* Outputs: return(int): the lag corresponding to the best match. The lag is
* with respect to the beginning of the history buffer.
*
* Date: 03-18-2009

***************************************************************************/
int PatternMatch(short *y)
{
    int   n;
    float maxCn;
    float Cn;
    int   bestmatch;
```

```
    maxCn=-999999.0;  /* large negative number */
    bestmatch=0;
    for (n=0;n<N;n++)
    {
       Cn = CrossCorrelation(&y[LHIST-M] /* x */, &y[n]);
       if (Cn>maxCn)
       {
          bestmatch=n;
          maxCn = Cn;
       }
    }
    return(bestmatch);
}

/*****************************************************************************
 * Function: AmplitudeMatch()
 *
 * Purpose: Perform amplitude matching using mean-absolute-value according
 *     to Goodman paper.
 *
 * Inputs:  *y : pointer to history buffer
 *     bestmatch : value of the lag to the best match
 *
 * Outputs: return(float): scale factor
*
 * Date: 03-19-2009
*****************************************************************************/
float AmplitudeMatch(short *y, short bestmatch)
{
    int   i;
    float sumx;
    float sumy;
```

```
    float sf;

    sumx = 0.0;
    sumy = 0.000001f;
    for (i=0;i<FS;i++)
    {
       sumx += abs(y[LHIST-FS+i]);
       sumy += abs(y[bestmatch+i]);
    }
    sf = sumx/sumy;

    /* This is not in the paper, but limit the scaling factor to something
       reasonable to avoid creating artifacts */
    if (sf<0.75f) sf=0.75f;
    if (sf>1.2f) sf=1.2f;
    return(sf);
}
```

13 Appendix D: Quality Metrics

13.1 Audio levels

The Bluetooth Core Specification (Volume 2, Part B, Section 9, Audio) describes the CVSD codec for voice on the Bluetooth air interface. The Core Specification also provides general audio recommendations for narrowband voice. This appendix provides recommendations for the Wide Band Speech update of the Hands-Free Profile, and for the Super Wide Band Speech update of the Hands-Free Profile.

For both CVSD (narrow band) and the Wide Band Speech codecs, the full scale sine wave PCM data should meet the network level of +3dBm0 as previously specified for the CVSD codec.

For Super Wide Band Speech codecs, a full scale 6427 Hz sine wave PCM data signal should meet the network level of +3dBm0 as previously specified for the CVSD codec. Note that for lower bitrates, the LC3 utilizes a Long-Term Post Filter (LTPF) which may amplify the signal. The frequency 6427 Hz minimizes this effect.

The AG manufacturer should adjust the gain from the Bluetooth Reference Point (BTR) to the cellular network, as shown in the core specification. The HF is responsible to adjust the speech signal to the BTR, as shown in the core specification, to enable nominal speech level of -21dbm0 in the networks at given nominal sound pressure level of -4.7 dBPa at the mouth reference point of the HF as described in ITU-T P.1100.

dBm0: absolute power level in dBm referred to a point of zero relative level (0 dBr point)

A possible test setup is described in Figure 13.1.

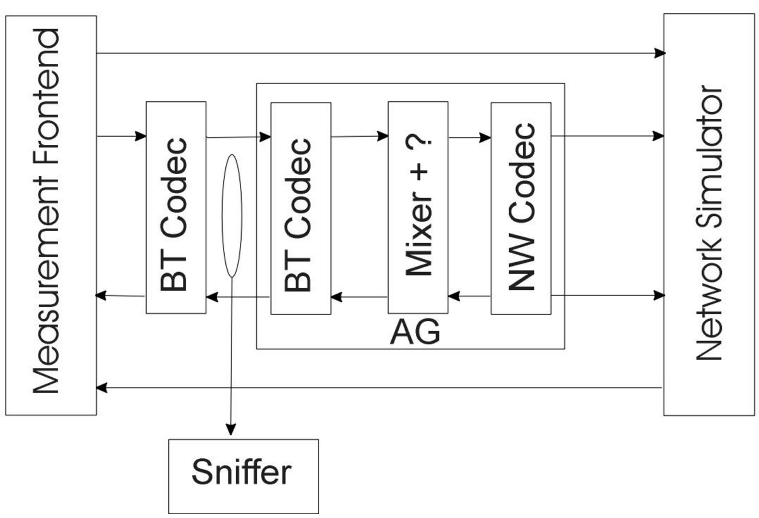

Figure 13.1: Example setup for audio level testing

The gain from BTR to the network and the gain from the network to the BTR should be adjusted for both signal processing modes: either on the phone (AG) or on the hands-free (HF) device after confirming the AT+NREC=0 command. When the signal processing is confirmed to be disabled on the AG, a sine signal can be used to verify the gain between the BTR and the network. When the active signal processing for Echo Cancellation and Noise Reduction are performed on the AG, a sine signal can lead to wrong results and a speech like signal, like P.50 (artificial voice), should be used. A Network simulator is highly recommended during testing because of the unknown status of gain and signal processing in real networks.

13.2 Bluetooth Sensitivity Frequency Responses

13.2.1 Bluetooth Send Sensitivity Frequency Response

The send sensitivity frequency response is measured from BTR to POI (reference speech codec of the system simulator, output).

The tolerance mask for the send sensitivity frequency response is enumerated in the Table below, the mask is drawn by straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale.

Frequency (Hz)	**Upper Limit**	**Lower Limit**
200	0	-2
5000	0	-2
12500	0	-4
14100 (14100 = 12500 +0.1*32000/2)	0	-5
16000	0	no lower limit at the Nyquist frequency

All sensitivity values for Super Wide Band Mask values are expressed in dB on an arbitrary scale.

Frequency (Hz)	**Upper Limit**	**Lower Limit**
100	0	-2
6 200	0	-2
7 000	0	-3

All sensitivity values for Wide Band Mask are expressed in dB on an arbitrary scale.

Frequency (Hz)	**Upper Limit**	**Lower Limit**
200	0	-2
3 100	0	-2
3 400	0	-3

All sensitivity values for Narrow Band Mask are expressed in dB on an arbitrary scale. Narrow band stops at 3400 Hz.

Table 13.1: Tolerance Mask for the Bluetooth Send Sensitivity Frequency Response

13.2.2 Bluetooth Receive Sensitivity Frequency Response

The receive sensitivity frequency response is measured from the electrical reference point (input of the system simulators, POI) to the Bluetooth reference interface.

The tolerance mask for the receive sensitivity frequency response is shown in the Table below, the mask is drawn by straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale.

Frequency (Hz)	Upper Limit	Lower Limit
200	0	-2
5000	0	-2
12500	0	-4
14100 (14100 = 12500 +0.1*32000/2)	0	-5
16000	0	no lower limit at the Nyquist frequency

All sensitivity values for Super Wide Band Mask values are expressed in dB on an arbitrary scale.

Frequency (Hz)	Upper Limit	Lower Limit
100	0	-2
6 200	0	-2
7 000	0	-3

All sensitivity values for Wide Band Mask are expressed in dB on an arbitrary scale.

Frequency (Hz)	Upper Limit	Lower Limit
200	0	-2
3 100	0	-2
3 400	0	-3

All sensitivity values for Narrow Band Mask are expressed in dB on an arbitrary scale. Narrow band stops at 3400 Hz.

Table 13.2: Tolerance Mask for the Receive Sensitivity Frequency Response

14 Appendix E: Technical Specification of LC3-SWB

14.1 Introduction

This appendix describes the additions for defining the LC3-SWB codec. The additions are limited to LC3-Codec ID and the LC3 parameter setup. All other parts of the LC3 definition remain unmodified.

14.2 Codec ID definition

LC3-SWB does not employ any additional frame header compared to mSBC.

Table 14.1 represents the values of the LC3-SWB parameters that correspond to the values specified in the Codec ID field. The Codec ID is used with the AT+BCS command, see Section 4.11.3.

8 bit Codec ID	**Mandatory or Optional**	**LC3 Parameters**					
		Channel mode	Sampling rate	byte_count	Bit depth, bits per audio sample	Frame duration	Header type
0x03	M	Mono	32 kHz	58	16	7.5 ms	H2

Table 14.1: Mapping of LC3-SWB Codec ID to LC3 parameters

14.3 Padding

LC3-SWB does not employ any padding.

14.4 Handling of eSCO 16-bit CRC

The erroneous data reporting feature (Volume 2, Part B in [1]) for the HCI Synchronous Data packets shall indicate lost data in the Packet Status flag.