Proposal for a dial setup command to two entities, revision 2, 01.07.2009

The system consists of a client that sends a command to a server in order to make the server set up a connection between two entities.
SIP framework with UDP transport layer is used for the transport of the command. Because of the three way handshake, the SIP INVITE transaction is used. The dial command is coded into a new defined SIP header that is included in the INVITE request. Only the client of this system can start a transaction. As in standard SIP INVITE transaction, there can be zero, one or more provisional responses from the server, but there is exactly one final response. For sepearating provisonal from final responses, the dedicated status numbers of standard SIP responses are used. The standard SIP message repitition timing is used. For authentification purposes, the normal SIP procedure will take place.
Unlike normal SIP INVITE transaction, this one with the dial command does not start a SIP dialog. In order to be able to work with standard SIP framework, no positive responses (2xx) are sent back by the server. Standard SIP framework will recognise the dialog not being initated.
Like in normal SIP framework, the client can terminate an INVITE transaction and abort the command execution with a CANCEL request. For correctly finishing the INVITE transaction, the client must acknowledge the final result code with sending of an ACK request.
Other requests than INVITE, ACK or CANCEL are not defined in this document and are not used in this system. If the client wants to ping the server without any actions, it can start an INVITE transaction without dial command header. The server will get back a negative final result code.
After the INVITE transaction is successfully or successless completed, the client has no further facilities to control the call between the two entities.

Header for command coding

The header name is 'AS55XDialCommand' ,all header-vaules of this header are comma separated. Header values consist of a field name, a colon ':', and a field value. Mandatory header values must be present, if optional values are missing, the default field values are used. The following header values are defined.

Field name:	'Number1'
Field value:	Numeric String of 1 to 21 digits, the first character may be '+'
Priority:	Mandatory

Field name:	'Number2'
Field value:	Numeric String of 1 to 21 digits, the first character may be '+'
Priority:	Mandatory

Field name:	'RoutingOption1'
Possible field values:	'ExclusivelyWireless' 'PreferablyWireless' 'ExclusivelyWired'
Priority:	Optional, default 'PreferablyWireless'

Field name:	'RoutingOption2'
Possible field values:	'ExclusivelyWireless' 'PreferablyWireless' 'ExclusivelyWired'
Priority:	Optional, default 'PreferablyWireless'

Field name: 'SequenceOption'
Possible arguments: 'CallNumber1First'
'CallSimultaneously'
Priority: Optional, default 'CallNumber1First'

Relevance of standard SIP headers and message bodies

For the command exchange the presence of the following standard SIP headers are mandatory: 'Via', 'To', 'From', 'CSeq', 'Call-ID'. Other standard or non-standard SIP haeaders may be included, but they are ignored by the server.
The calling and called party numbers in the From and To headers, in the request line and - if included - in the Contact header are irrelevant and should be fixed to '0'. The building of the SIP URI must accord to standard SIP convention.
A message body may be attached, but it is ignored by the server.

Coding of responses

The server is using the following response values:
'100 Trying' as provisonal response if the command is accepted
'183 Session Progress (reason phrase) if one entity is ringing or has acceted the call
The following reason phrases for response 183 are defined:
'Entity1Ringing'
'Entity1Accepted'
'Entity2Ringing'
'Entity2Accepted'
'410 Gone (reason phrase)' as final response when the command has been executed
The following reason phrases for response 410 are defined:
'Success' if the command was successfully executed
'CommandHeaderMissing' if the command header is not included
'CommandSyntaxError' if an error in the command header is detected
'Entity1Busy'
'Entity2Busy'
'Entity1NotReachable'
'Entity2NotReachable'
'401 Unauthorized' if authentification is necessary

Server behavior

If the server receives a dial command, it will check the command and send back a TRYING provisional response if the command is accepted. If the command cannot be executed, the transaction is immediately terminated with a final response. If the command has been accepted and the execution fails or if the command execution succeeded, the transaction is terminated with a final response as well. The server will send further provisional responses, if one entity is ringing or has accepted the call. In worst case, the execution time can reach up to about five minutes (two times maximum dial-up times and both entities accept call after maximum ringing time).
If the server receives a CANCEL request during the command execution, it terminates execution and closes the correspondend INVITE transaction with a final response.
If the final result code reason phrase is not 'Success', possibly startet call setups or already present calls to the entities are already termianted or disconnected.

Client behavior

If the client does not receive a final response within five minutes, it should terminate the INVITE transaction with a CANCEL request. All other sequence and timing rules are standard SIP.

Examples

Only mandatory SIP headers are shown, the client IP address is 192.168.100.236 and the server IP address is 192.168.100.235.

Example 1: Normal successful call setup

Client sends ...
INVITE sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
AS55XDialCommand:
Number1:+123456789,RoutingOption1:ExclusivelyWireless,Number2:+123456780,RoutingOption2:
ExclusivelyWired,SequenceOption:CallNumber1First
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Server sends ...
SIP/2.0 100 Trying
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Server sends ...
SIP/2.0 183 SessionProgress (Entity1Ringing)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Server sends ...
SIP/2.0 183 SessionProgress (Entity1Accepted)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Server sends ...
SIP/2.0 183 SessionProgress (Entity2Ringing)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Server sends ...
SIP/2.0 410 Gone (Success)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 INVITE

Client sends ...
ACK sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabc@192.168.100.236
CSeq: 653196842 ACK

Example 2: Wrong dial command

Client sends ...
INVITE sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
AS55XDialCommand:
Number1:+123456789,RoutingOption1:ExclusivelyWireless,RoutingOption2:ExclusivelyWired,SequenceOption:CallNumber1First
Call-ID: 96f36ef5630dabd@192.168.100.236
CSeq: 653196843 INVITE

Server sends ...
SIP/2.0 410 Gone (CommandSyntaxError)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabd@192.168.100.236
CSeq: 653196843 INVITE

Client sends ...
ACK sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabd@192.168.100.236
CSeq: 653196843 ACK

Example 3: Entity 2 is busy

Client sends ...
INVITE sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
AS55XDialCommand:
Number1:+123456789,RoutingOption1:ExclusivelyWireless,Number2:+123456780,RoutingOption2:
ExclusivelyWired,SequenceOption:CallNumber1First
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 INVITE

Server sends ...
SIP/2.0 100 Trying
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 INVITE

Server sends ...
SIP/2.0 183 SessionProgress (Entity1Ringing)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 INVITE

Server sends ...
SIP/2.0 183 SessionProgress (Entity1Accepted)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 INVITE

Server sends ...
SIP/2.0 410 Gone (Entity2Busy)
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 INVITE
Client sends ...
ACK sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabe@192.168.100.236
CSeq: 653196844 ACK

Example 4: Authorisation required
Note: {...} are variable values of standard SIP framework

Client sends ...
INVITE sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
AS55XDialCommand:
Number1:+123456789,RoutingOption1:ExclusivelyWireless,Number2:+123456780,RoutingOption2:
ExclusivelyWired,SequenceOption:CallNumber1First
Call-ID: 96f36ef5630dabf@192.168.100.236
CSeq: 653196847 INVITE

Server sends ...
SIP/2.0 401 Unauthorized
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabef@192.168.100.236
WWW-Authenticate: {...}
CSeq: 653196847 INVITE

Client sends ...
ACK sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235;tag=6845645136
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
Call-ID: 96f36ef5630dabf@192.168.100.236
CSeq: 653196847 ACK

Client sends ...
INVITE sip:0@192.168.100.235:5060 SIP/2.0
From: sip:0@192.168.100.236;tag=844168c63d8f68c
To: sip:0@192.168.100.235
Via: SIP/2.0/UDP 192.168.1.2:5060;branch=z9hG4bK4cfb93b64
AS55XDialCommand:
Number1:+123456789,RoutingOption1:ExclusivelyWireless,Number2:+123456780,RoutingOption2:
ExclusivelyWired,SequenceOption:CallNumber1First
Call-ID: 96f36ef5630dabf@192.168.100.236
CSeq: 653196848 INVITE
Authorization: {...}

Server sends ...
SIP/2.0 100 Trying
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