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Analysis of Middlebox Interactions for Signaling Protocol Communication along the Media Path
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Abstract

Middleboxes are defined as any intermediary box performing functions apart from normal, standard functions of an IP router on the data path between a source host and destination host. Two such functions are network address translation and firewalling.

When Application Layer Gateways, such as SIP entities, interact with NATs and firewalls, as described in the MIDCOM architecture, then problems may occur in the transport of media traffic when signaling protocol interaction takes place along the media path, as it is the case for recent key exchange proposals (such as DTLS-SRTP). This document highlights problems that may arise. Unfortunately, it is difficult for the end points to detect or predict problematic behavior and to determine whether the media path is reliably available for packet exchange.

This document aims to summarize the various sources and effects of NAT and firewall control, the reasons that they exist, and possible means of improving their behavior to allow protocols that rely upon signaling along the media path to operate effectively.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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1. Introduction

According to by RFC 3234 [RFC3234] middleboxes are defined as any intermediary box performing functions apart from normal, standard functions of an IP router on the data path between a source host and destination host.

In the context of SIP a SIP ALG may interact with a node along the media path to control network address translation, firewalling, and other functions.

> With firewall control packet filters are installed based on the SIP signaling interaction to implement a behavior of 'deny by default' in order to reduce the risk of unwanted traffic. This function is often referred to as 'gating'. Depending on the timing of the packet filter installation and the content of the packet filter signaling traffic along the media, such as DTLS-SRTP or ICE, may be treated in an unexpected way.
>
> In cases where the middlebox is involved in overcoming unmanaged NAT traversal the case is similar. The key feature of this type of NAT traversal is a desire to overcome the possible lack of information about any [RFC4787] address and/or port mapping by a possibly unknown NAT device (server reflexive address and filtering properties). In particular, a NAT binding for an endpoint may not exist yet for the address and port identified in the endpoint's SDP. As such, a pilot packet sent by that endpoint behind the NAT is required to create the necessary mappings in the NAT for the media relay to deliver media destined for that endpoint. Until that pilot packet is received no media packets may be reliably forwarded to the endpoint by the relay.

This document presents a summary of these two techniques, discusses their impact upon other protocols such as ICE and DTLS-SRTP, and proposes a set of recommendations to mitigate the effects of gating and latching on in-band negotiation mechanisms.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

We use the terms filter, policy action (or action), policy rule(s), MIDCOM agent, and MIDCOM Policy Decision Point (PDP) as defined in [RFC3303]. The MIDCOM agent is co-located with a SIP ALG that communicates with the firewall or the media relay.

3. Architecture

Figure 1 shows the architecture that is being considered in this document with respect to firewall and NAT traversal using media relaying. The timing and directionality with which media packets are allowed to traverse a particular edge device is the subject of this investigation. The MIDCOM agent thereby pushes policy rules to the middlebox that allow or deny certain flows to pass through. Additionally, in case of media relaying it is important for the MIDCOM agent to adjust the signaling messages.

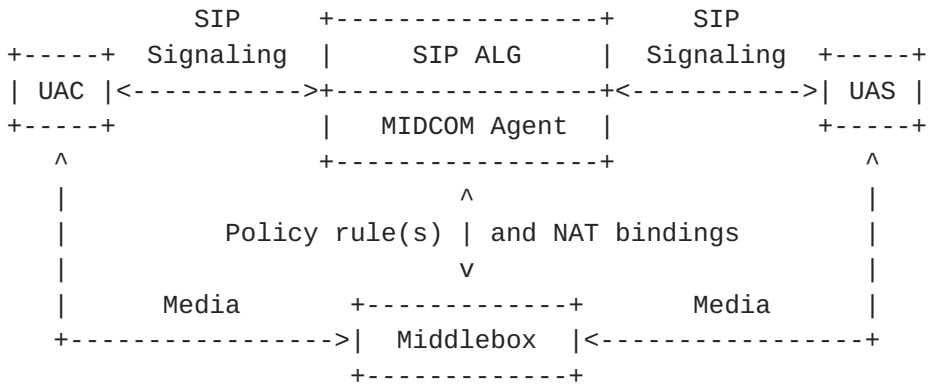


Figure 1: Analysed Firewalling Architecture

The aspects of packet filtering are described in Section 4 whereas NAT traversal is illustrated in Section 5.

4. Packet Filtering

Figure 1 highlights the interaction between the MIDCOM agent and the middlebox. These two elements inspect call control signaling and media path packets and determine when packets from a given source to a given destination are allowed to flow between endpoints. It is common for the gate controller to be the local outbound proxy for a given SIP UA being gated.

The primary responsibility of the MIDCOM agent, which is co-located with a SIP entity, is to examine the call control signaling to determine the media addresses and ports used to define the media path between the gated device and the endpoint(s) with which it is corresponding. For SIP, this would correspond to the media addresses described within SDP after at least one full offer/answer exchange.

This information is used to create one or more packet filters that describe the expected media path(s) for the call. These packet filters are combined with an algorithmic determination, typically based on the state of the call, as to which direction(s) media

packets are allowed to flow between the endpoints, if at all. The filter and the action that is being installed by the MIDCOM agent at the middlebox may change during the lifetime of a SIP signaling session, depending on the state of the call or on changes of the address and port information of one (or even both) of the end points.

It is possible that the gate controller may not be able to establish an exact address or port for one endpoint involved in the call in which case it may wildcard the address and/or port for the source and /or destination endpoint in the packet flow filter. In such a case, the packet flow filter is considered to have matched against a given media packet for the wildcarded field.

Note that it is possible to specify the filter using wildcards, for example, if some end point address information is not known at a given point in time. Additionally, the default firewalling policy is subject to local configuration ('deny per default' vs. 'permit per default'). For a given SIP signaling sessions the policy at the MIDCOM agent might be very strict with respect to the packets that are allowed to flow in a particular direction. For example, packets may be allowed to flow in both directions, only in one direction for a specific media stream. No particular behavior can be assumed.

When a media session is destroyed (end of call, deleted from the session description, etc.), the MIDCOM agent removes policy rules created for that media session at the middlebox.

4.1. Protocol Interaction

MIDCOM agents may employ a variety of models to determine when to change the status of a particular policy rule. This is especially true when a call is being established. For SIP, this would be when an early dialog is established between endpoints. Although there is the potential for a great deal of variability due to an intentional lack of specification, typically, one of two models is used by the MIDCOM agent to determine the state of a policy rule during call setup: single-stage and two-stage commit. The term 'commit' here refers to the point at which a policy rule is setup that allows media traffic to flow. For example, this would be the point at which packets for a media stream marked a=sendrecv in SDP was allowed to flow bi-directionally by the middlebox.

4.1.1. Single-Stage Commit

Single stage commit is commonly used when the MIDCOM agent is most involved only in firewalling. For SIP, MIDCOM agents use a single-stage commit model typically install policy rules for the call when the 200 OK to the INVITE is received in the case that the INVITE

contained an SDP offer, or when the ACK is received if the initial offer was sent in the 200 OK itself.

This model is often used to prevent media from being sent end-to-end prior to the call being established.

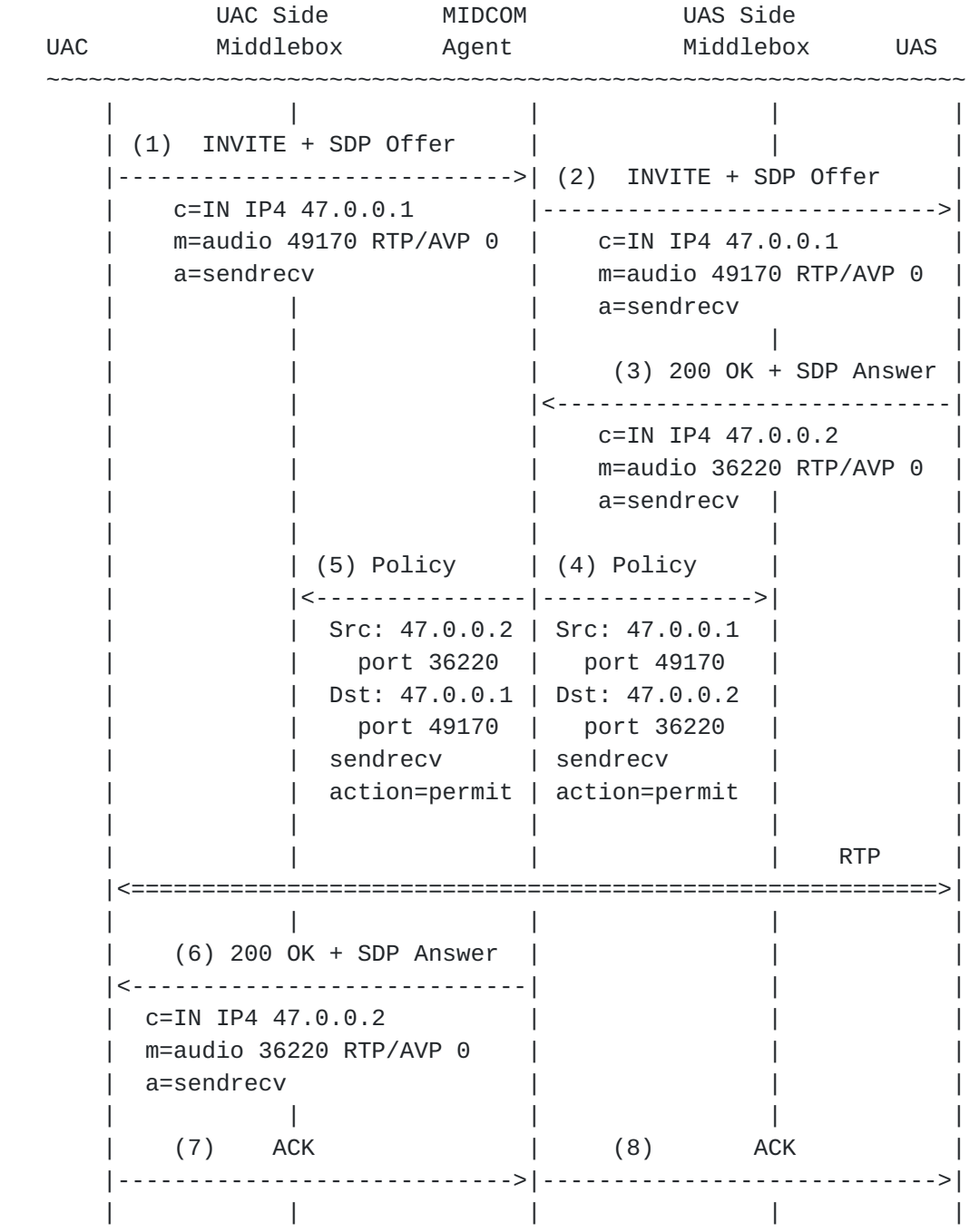


Figure 2: Example Single-stage Commit with SIP and SDP

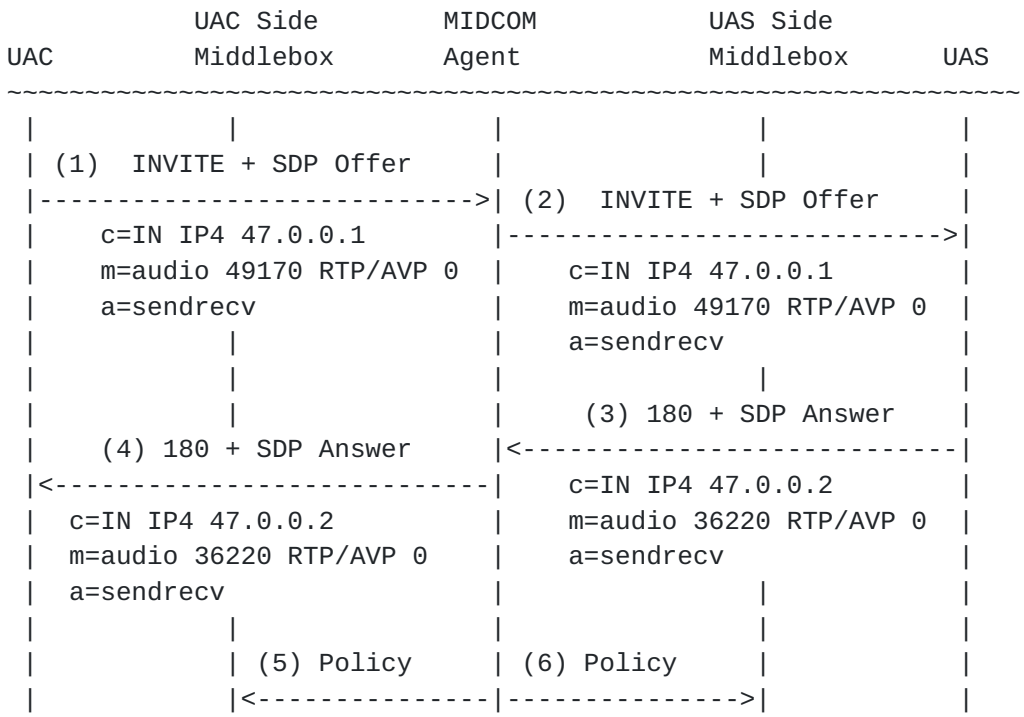
In the example above, policy is created in steps 4 and 5 to allow bi-directional media flow based on the SDP exchanged in steps 1 and 3. In particular, the rules at the UAC side middlebox would indicate that traffic exchanged between IP address 47.0.0.1 and port number 49170 and IP address 47.0.0.2 and port number 36220 is allowed in both directions.

In this example, the MIDCOM agent installs the policies after the 200 OK to the INVITE arrives in step 3. With a firewalling policy of 'deny by default' media sent prior to steps 5 and 4 by the UAC or UAS is discarded by the middleboxes.

Noted that early media that arrives before the 200 OK would require special treatment since otherwise it would be dropped as well.

4.1.2. Two-Stage Commit

Two-stage commit is used when the MIDCOM agent also providers functionality, such as Quality of Service signaling that may require resources to be reserved early on in the call establishment process before it is known if the call will be answered. An example of this would be where the MIDCOM agent is responsible for guaranteeing a minimum level of bandwidth along the media path. In this case an initial set of policies may be sent by the MIDCOM agent to the middlebox even though they are put into a pending state but trigger a resource reservation. Later, when the call is accepted, the gate controller may update the state of the policies to active them.



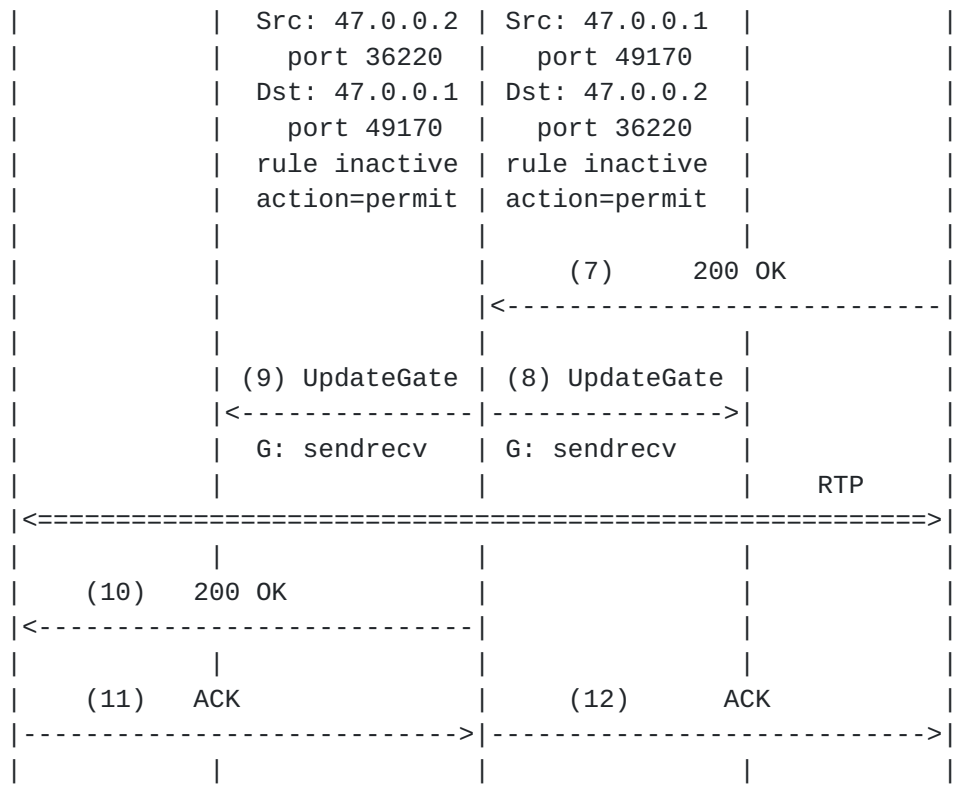


Figure 3: Example Two-stage Commit with SIP and SDP

In the example above, policies are created in steps 5 and 6 based off of the SDP sent in steps 1 and 3 in an initial inactive state (no packets are allowed to flow) despite the SDP indicating the media should be bi-directional. This interaction with the middlebox, however, triggers a QoS reservation to take place. Later, when the 200 OK to the INVITE comes in step 7, the policies are updated in steps 8 and 9 to indicate that packets should be allowed to flow bi-directionally. Although functionally equivalent to the single-stage commit example given earlier in Figure 2, other operations at the gate agent may have been performed simultaneously in steps 5 and 6 that justifies the early explicit definition of the gates in an inactive state. The full usage of PRACK here is not shown for purposes of brevity.

4.2. Further Reading

Packet filtering based on the approach described in this document has been described in a number of documents. Although the usage of this architecture can also be found on the Internet their behavior is largely specified only in documents that relate to IMS standardization. The behavior of the devices deployed on the Internet is therefore largely undocumented. Nevertheless, the following documents give the reader a better idea of the

functionality and the signaling interaction. These documents may also specify an additional behavior in relation to how packet filtering is used when the MIDCOM agent is responsible for processing SIP/SDP call control signaling and the middlebox is responsible for a variety of activities beyond pure filtering. For example, it is common for middleboxes to exempt RTCP flows from being blocked even though the associated RTP flows are not allowed to flow in order to support RTCP signaling while a call is on hold. These references are given here for the reader to gather a better understanding of how this is mechanism is used in various forums and is non-exhaustive:

1. 3GPP, "TS 23.203: Policy and charging control architecture" [TS-23.203]

2. 3GPP, "TS 29.212: Policy and Charging Control over Gx reference point" [TS-29.212]

3. 3GPP, "TS 29.213: Policy and Charging Control signaling flows and QoS parameter mapping" [TS-29.213]

4. 3GPP, "TS 29.214: Policy and charging control over Rx reference point" [TS-29.214]

5. ETSI TISPAN, "ES 282-003: Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control Sub-system (RACS); Functional Architecture" [TISPAN-ES-282-003]

6. Cablelabs, "PacketCable 2.0: Quality of Service Specification (PKT-SP-QOS-I01-070925)" [PKT-SP-QOS-I01-070925]

Note that different terms are used for the MIDCOM agent and the middlebox. For example, in an IMS context the MIDCOM agent would be part of the P-CSCF and PCRF elements or in TISPAN it would be part of the P-CSCF, A-RACF and SPDF that are involved in controlling gating operations. Many different elements perform the role of a middlebox: GSM GGSN, CDMA PDSN, EPC PDN Gateway, TISPAN RCEF and C-BGF/I-BGF, PacketCable CMTS, etc. These functions may be present in the network in a unified or decomposed architecture.

5. NAT Traversal

Two distinct types of NAT traversal can be supported by a MIDCOM agent and the connected middlebox:

1. The MIDCOM agent and the attached middlebox act as a B2BUA at the border of an operator's network to protect this network and to perform the IP address and port conversion, which may be required

because private address spaces are used within the network, or because IPv4 and IPv6 address realms are interfacing. For this use case, the middlebox itself performs functions similar to a NAT and is deployed instead of a NAT at a network border.

2. The MIDCOM agent and attached middlebox support the traversal of a residential NAT (also termed customer premise equipment), which is typically located at the user's side of an access network, for instance within a DSL router. The middlebox thereby acts as kind of media relay.

Both functions can be combined by the same MIDCOM agent and connected middlebox, for instance by a TISPAN C-BGF.

As shown in Figure 1 the MIDCOM agent that is being co-located with the SIP ALG functionality interacts with the middlebox that is also a NAT in order to request and allocate NAT bindings and then modifies the SDP offer and answer within SIP to insert the IP addresses and port allocated by the NAT as destination for the media in both directions. A consequence of the interaction with a (double) NAT is that the media traffic is forced to traverse a certain NAT in both directions (also called media anchoring). The opening of pinholes through the middlebox is only done on request of the MIDCOM agent, and not triggered by the detection of outbound media flows. Such middleboxes are for instance the TISPAN 3GPP Tr-GW/C-BGF/I-BGF and the 3GPP IMS Access Gateway.

The functionality and control of the middlebox becomes comparable to a media gateway and TISPAN standardized the usage of the H.248 / MEGACO protocol for the control of the middlebox by the midcom MIDCOM agent.

This architecture could be compared with a STUN relay [RFC5766] that is being controlled by the MIDCOM agent rather than the end point itself. The motivation why this technique is being used in favor to other NAT traversal techniques is that clients do not have to support anything beyond RFC 3261 [RFC3261] and network administrators can control and apply local policy to the relay binding process in a centralized manner.

5.1. Protocol Interaction

The MIDCOM agent's role is to inspect call control signaling and update media address and port values based upon media relay binding information allocated with the middlebox/media relay. For SIP, this minimally involves updating the c= and m= lines in the SDP, although some implementations may also update other elements of the SDP for various reasons.

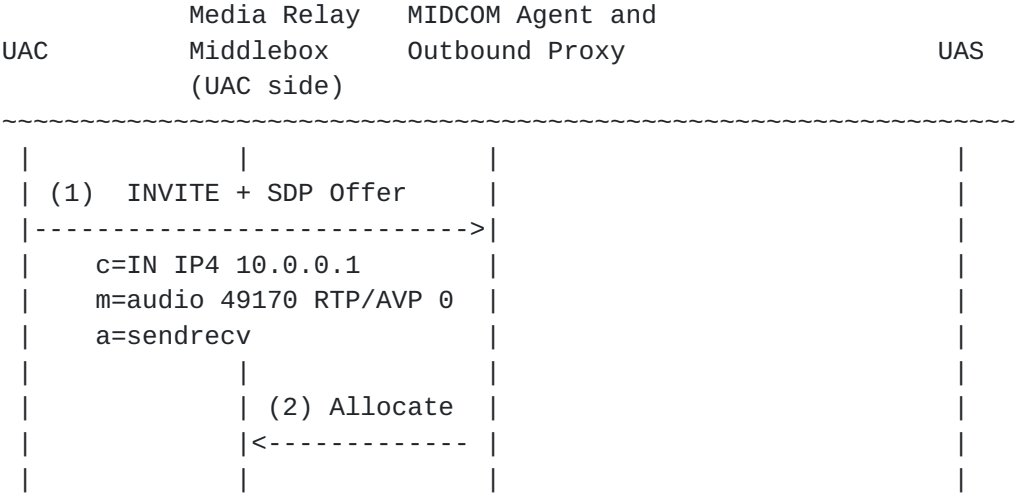
Because the endpoints may not be able to gather a server reflexive address for their media streams, the MIDCOM agent employs the following algorithm to ensure that media can flow to the given endpoint:

1. When receiving an initial SDP offer, the MIDCOM agent requests authorization for the request arriving at the middlebox, configures the middlebox to forward media between the offerer and the destination address / port as received in the incoming SDP offer, reserves a local IP address and port, and replaces the destination address and port from the incoming offer with the IP address / port used by the middlebox in the forwarded offer.

2. When receiving an initial SDP answer, the MIDCOM agent configures the middlebox for the corresponding session to send media towards the answerer towards the destination address and port as received in the incoming SDP answer, request the middlebox to reserve a local IP address / port, and exchange the destination address and port from the incoming answer with that middlebox IP address and port in the forwarded answer.

3. If the middlebox supports the traversal of residential NATs, it applies a technique called "media latching": The destination IP address of packets forwarded by the middlebox in the outbound direction is derived from the source IP address of packets received in the inbound direction. This overrides a destination address possibly configured by the MIDCOM agent.

An example of this algorithm is shown in Figure 4 when using SIP and SDP. In this example the UAC is the endpoint served by the MIDCOM agent, which is also acting as a local outbound proxy, and the UAS is the corresponding endpoint. We assume that the UAC is located behind a residential NAT; this NAT is, however, not shown in Figure 4.



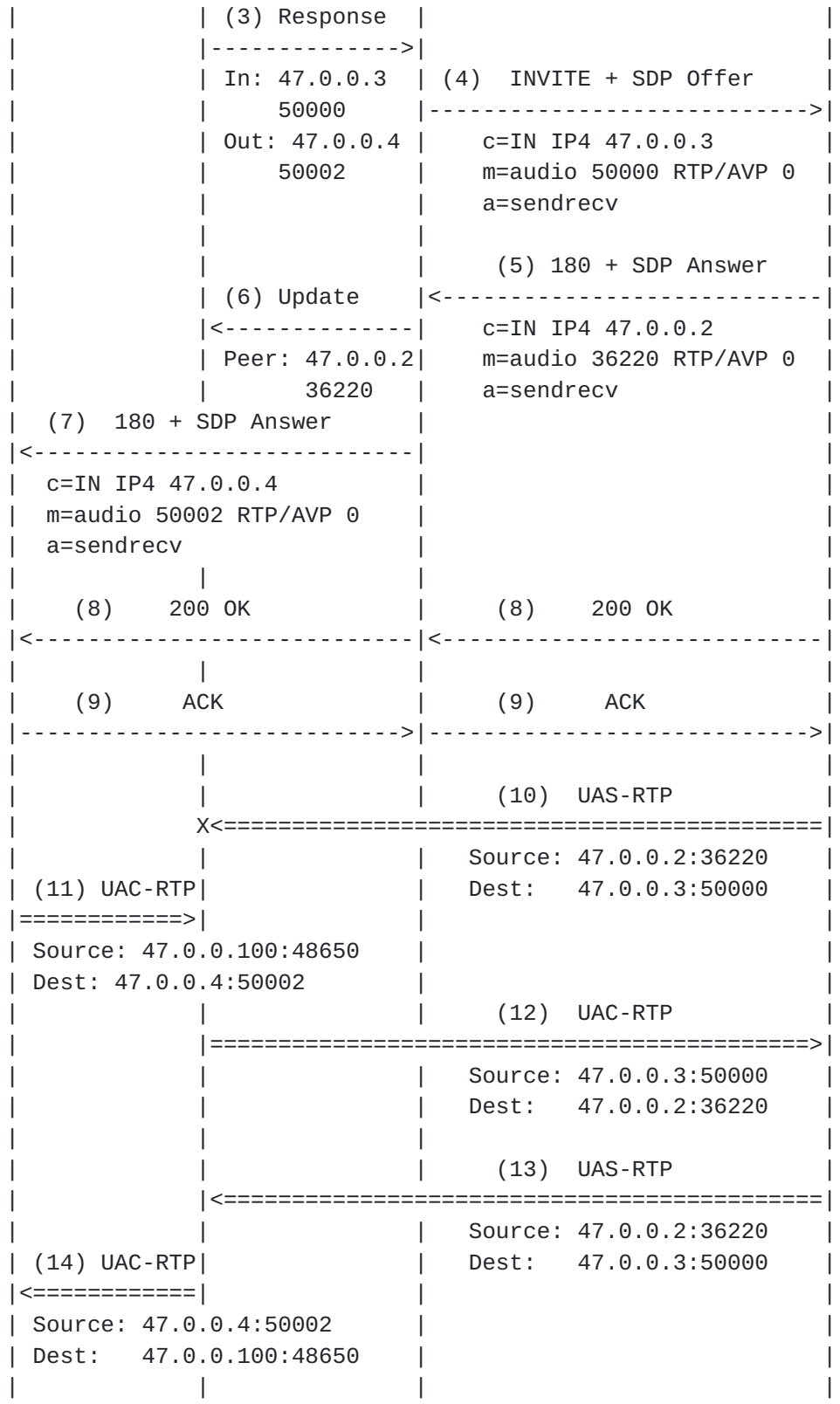


Figure 4: Call Flow with SIP + SDP

Step (1): UAC sends INVITE to local outbound proxy, which is also a MIDCOM agent, with an SDP offer.

Step (2): The MIDCOM agent looks at the signaling and asks the middlebox to allocate a media relay binding. At this point in time the MIDCOM agent can only provide the IP address it finds inside the offer, i.e., the IP address and port where the UAC is expecting to receive traffic sent by the UAS. In this example the IP address equals 10.0.0.1 and the port number is 49170.

Step (3): The middlebox responds with a media relay binding that consists of an inbound address/port for media sent by the UAS, and an outbound address/port for media sent by the UAC. The IP address and port of the middlebox allocated for the inbound side 47.0.0.3:50000 and the address and port on the outbound side is 47.0.0.4:50002.

Step (4): The MIDCOM agent updates the addresses in the SDP offer with the inbound address/port information from the middlebox/media relay binding response, namely with 47.0.0.3:50000.

Step (5): The UAS responds with a 180 containing an SDP answer. This answer indicates that traffic will be sent from the IP address and port 47.0.0.2:36220.

Step (6): The MIDCOM agent interacts with the middlebox to update the destination address/port information from the SDP answer for media to be sent to the UAS, and changes the addresses/ports in the SDP answer to the UAC with the outbound address/port information from the middlebox binding from step 3. Media can now flow to the UAS from the UAC at the middlebox/media relay, i.e., in the outbound direction.

Step (7): The UAC receives the SDP answer containing the media relay outbound address/port information, namely 47.0.0.4:50002.

Step (8): The UAS answers the INVITE with a 200 OK.

Step (9): The UAC acknowledges with an ACK.

Step (10): RTP for the UAS, which may have begun flowing prior to answer, goes to the middlebox, but the middlebox has no reliable address to relay the media to for the UAC yet. Media will typically be dropped.

Step (11): RTP arrives at the media relay on the inbound address/ port from the UAC. The middlebox observes the source address and port of the arriving packet and completes the binding process.

The source address and port of the media from the UAC is now the destination address/port for media arriving on the outbound port of the middlebox/media relay from the UAS.

Step (12): Media originating from the UAC is relayed by the middlebox to the UAS.

Step (13): Media from the UAS is sent towards the middlebox.

Step (14): The middlebox forwards the media traffic to the UAC.

5.2. Further Reading

In TS 23.228 the 3GPP standardized the usage of a SIP-ALG residing in the P-CSCF to control an IMS Access Gateway, acting as middlebox at the interface between the IMS and the access network (see Annex G), and the usage of a SIP-ALG residing in the IBCF to control an TrGW as a middlebox at the interface between the IMS and external networks or other IMS networks (see Annex I).

Although the described residential NAT traversal approach is used by a number of implementations to overcome incorrect address/port information in call control signaling from an endpoint behind a NAT, only one reference is known that describes the functionality in a standardized manner.

1. ETSI TISPAN, "ES 282-003: Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control Sub-system (RACS); Functional Architecture" [TISPAN-ES-282-003]. The TISPAN Ia interface between the TISPAN BGF and SPDF is the relevant specification.

6. Interactions between Media Path Signaling and Middlebox Behavior

This section points to the problems that occur when signaling exchanges are performed along the media path when middleboxes are present that behave in the way described in this document.

6.1. Packet Filtering

The description in Section 4 highlighted that the timing of the policy rule installation by the MIDCOM agent towards the middlebox has an impact on when and what media traffic is allowed to traverse.

The installation of policy rules is a prerequisite for related media to flow. As those policy rules are derived from information from both SDP offer and answer, they are typically installed at the completion of the first offer-answer exchange.

Furthermore, the middlebox may prevent the exchange of packets in the media path after this point by closing "gates" until the session establishment signaling has reached a pre-configured milestone where the MIDCOM agent signals to the middlebox that packets are allowed to traverse in both directions. Prior to this, packets may be allowed to flow uni-directionally to satisfy certain service requirements or may be entirely blocked by the middlebox. For SIP [RFC3261] the typical milestone that must be reached is offer/answer exchange [RFC3264] accompanied by an acknowledgement that the dialog has been accepted by the UAS (i.e., 200 OK to the INVITE). It depends on the policy of an operator when to open gates. The policy may take into account the requirements of special media types to have early bidirectional media exchanges, e.g. if the usage of DTLS is indicated in SDP.

A concrete example of the impact can be found with the case of key exchange along the media path, as it is provided by DTLS-SRTP. The ladder diagram in Section 7.1 of [RFC5763] shows that the arrival of the SIP INVITE at the UAS triggers the DTLS handshake. This message would be blocked by the middlebox, as described in Section 4 since the MIDCOM agent has not yet installed policy rules. The consequence is that the communication fails unless the UAS repeats attempts for an DTLS handshake until connectivity is established in both directions by the installation of policy rules and the presence of opened gates. Due to extra time required for the DTLS exchange the user may experience clipping.

According to 3GPP standards, gates for RTCP are always opened when policy rules for related media are installed, even if related media traffic is still blocked. Therefore, signaling embedded in RTCP is likely to pass after the completion of the first offer-answer exchange. Standardized policy rules only inspect source and destination information of IP packets and the transport protocol (e.g., UDP and TCP). Obviously, this is not a property that can be guaranteed to be true in the future.

6.2. NAT Traversal

The described NAT traversal interaction prevents asynchronous exchange of packets in the media path until a pilot packet has been received by the middlebox from the endpoint being served. It can be employed for both the [RFC3264] offerer and/or answerer. Therefore, in the worst case, both endpoints must generate a pilot packet

towards each other to ensure a bi-directional media path exists. Any signaling on the media path that relies upon a uni-directional handshake in the reverse direction may not complete until media in the forward direction by the other endpoint. If signaling on the media path is required to complete prior to media generation the handshake may stall indefinitely.

Middleboxes as described in Section 5 will not allow any media to pass through without being configured to do so by the MIDCOM agent when the first offer-answer exchange is completed. Without latching, it may be technically feasible to pass media packets from answerer towards the offerer after the offer has passed the MIDCOM agent, but existing implementations hardly show that behavior. Furthermore, such middleboxes may apply gating policies similar to the policies discussed in Section 6.1 in addition.

The described latching technique for residential NAT traversal interaction requires that a pilot packet has been received by the middlebox from the endpoint being served before the middlebox is able to send packets towards the endpoint. This latching technique can be employed for both the RFC 3264 offerer and answerer. Therefore, in the worst case, both endpoints must generate a pilot packet towards each other to ensure that a bi-directional media path exists. If the first packets to be exchanged in the media path are signaling packets and a particular directionality of those packets is required, communication may fail. To overcome these problems, empty packets could be sent by the endpoint that has to receive rather than to send the first signaling message. The offer is capable of sending the pilot packet only when receiving the destination information within the answer. Thus, before that point in time the offerer will also not be able to receive any media packets or related signaling.

In a similar manner as outlined in Section 6.1, any in-path signaling messages that are sent before the offer-answer exchange is completed will be dropped.

7. Preliminary Recommendations

The following preliminary recommendations are suggested:

REC #1: It is recommended that any protocol handshake on the media path ensure that a mechanism exists that causes both endpoints to send at least one packet in the forward direction as part of, or prior to, the handshake process. Retransmission of STUN connectivity checks (see [RFC5389]) as part of ICE [RFC5245] is an example of such a mechanism that satisfies this recommendation. Sending of no-op RTP packets (see [I-D.ietf-avt-rtp-no-op]) is another example.

REC #2: It is recommended that middleboxes present on the media path allow at least a nominal amount of traffic to be exchanged between endpoints after the completion of the first offer-answer exchange to enable the completion of media path signaling prior to the session being established. Such policies may be restricted to media types that use in-path signaling. The amount of traffic necessary to complete the signaling between endpoints is expected to be orders of magnitude smaller than that of any sufficiently interesting fraudulent traffic.

REC #3: It is recommended that failure to complete signaling on the media path not automatically cause the session establishment to fail unless explicitly specified by one or more endpoints. A fallback scenario where endpoints retry signaling on the media path is recommended. Recommended points in time to retry signaling on the media path are after the completion of the first offer-answer exchange and again after the session has been established. Additional retries with adequate pacing may be used in addition.

REC #4: If signaling on the media path is required before media can flow, the answerer should send the SDP answer as soon as possible, for example within a provisional SIP response, to allow the media path signaling to pass through middleboxes and therefore to avoid clipping.

REC #5: It is recommended that middleboxes present on the media path allow at least a nominal amount of traffic to be exchanged between endpoints for at least one RTT after the middlebox receives a message from the MIDCOM agent indicating the media session being terminated. This will ensure that any transit signaling packets on the media path exchanged during the session termination pass through the middlebox.

8. Security Considerations

This document talks about security related functionality and the impact of one security mechanism, namely firewalling, to another one, namely key management for media security.

9. IANA Considerations

This document does not require actions by IANA.

10. Acknowledgements

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11. References

11.1. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[RFC3264] Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, June 2002.

[RFC3303] Srisuresh, P., Kuthan, J., Rosenberg, J., Molitor, A., and A. Rayhan, "Middlebox communication architecture and framework", RFC 3303, August 2002.

11.2. Informative References

[I-D.ietf-avt-rtp-no-op] Andreasen, F., "A No-Op Payload Format for RTP", draft-ietf-avt-rtp-no-op-04 (work in progress), May 2007.

[PKT-SP-QOS-I01-070925] CableLabs, "PacketCable 2.0: Quality of Service Specification", September 2007, <http://www.cablelabs.com/specifications/PKT-SP-QOS-I01-070925.pdf>.

[RFC3234] Carpenter, B. and S. Brim, "Middleboxes: Taxonomy and Issues", RFC 3234, February 2002.

[RFC4347] Rescorla, E. and N. Modadugu, "Datagram Transport Layer Security", RFC 4347, April 2006.

[RFC4787] Audet, F. and C. Jennings, "Network Address Translation (NAT) Behavioral Requirements for Unicast UDP", BCP 127, RFC 4787, January 2007.

[RFC5245] Rosenberg, J., "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols", RFC 5245, April 2010.

[RFC5389] Rosenberg, J., Mahy, R., Matthews, P., and D. Wing, "Session Traversal Utilities for NAT (STUN)", RFC 5389, October 2008.

[RFC5763] Fischl, J., Tschofenig, H., and E. Rescorla, "Framework for Establishing a Secure Real-time Transport Protocol (SRTP) Security Context Using Datagram Transport Layer Security (DTLS)", RFC 5763, May 2010.

[RFC5766] Mahy, R., Matthews, P., and J. Rosenberg, "Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)", RFC 5766, April 2010.

[TISPAN-ES-282-003]
ETSI, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control Sub-system (RACS); Functional Architecture", June 2006, <http://webapp.etsi.org/>.

[TS-23.203]
3GPP, "Policy and charging control architecture", September 2007, <http://www.3gpp.org/ftp/Specs/html-info/23203.htm>.

[TS-29.212]
3GPP, "Policy and Charging Control over Gx reference point", June 2008, <http://www.3gpp.org/ftp/Specs/html-info/29212.htm>.

[TS-29.213]
3GPP, "Policy and Charging Control signaling flows and QoS parameter mapping", June 2008, <http://www.3gpp.org/ftp/Specs/html-info/29213.htm>.

[TS-29.214]

3GPP, "Policy and charging control over Rx reference point", June 2008, <http://www.3gpp.org/ftp/Specs/html-info/29214.htm>.

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