Non-Repudiation in Internet Telephony

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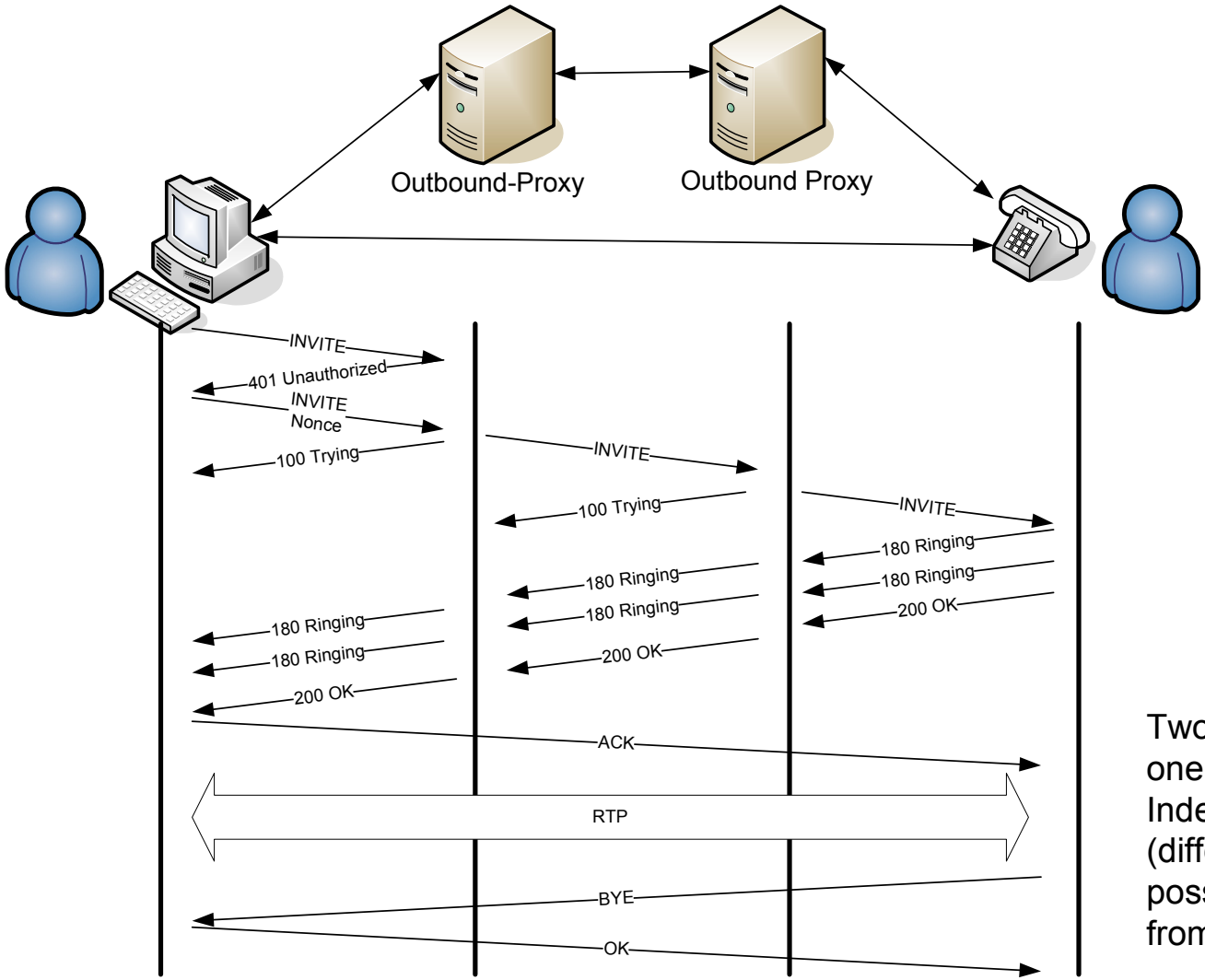
VoIP – a shift of paradigms in business communication

- VoIP is becoming the prevalent form of voice communication in commercial environments
  - Will carry more than 50% of business voice traffic in few years
  - 2007, over 60% of contact centres worldwide have switched to VoIP (up from 50% 06)
- Major reasons for switching to VoIP
  (Dimension Data‘s Global Contact Centre Benchmarking Report)
  - Flexibility of architecture
  - Cost savings
  - Improved business functionality
  - Replacement of end-of-life technology
- „Since contact centres depend on a range of I&C technologies, converged technology can significantly in crease efficiencies.“
- VoIP is gaining shares in mobile communication
- Industry quickly adopted the relevant standards SIP/RTP
  (Even Skype recently announced adopting them)

SIP – Session Initiation Protocol



Two RTP channels, one for each direction; Independent (different port and possibly IP numbers) from signalling

Incoming call example

```
INVITE sip:49601234567@84.178.139.124:5070 SIP/2.0
Record-Route: <sip:212.227.15.225;ftag=1168841412;lr=on>
Record-Route: <sip:+49601234567@217.188.44.231;ftag=1168841412;lr=on>
Via: SIP/2.0/UDP 212.227.15.197;branch=z9hG4bK679d78391b60eeef2b991fb1c5eef30f
Via: SIP/2.0/UDP 212.227.15.225;branch=z9hG4bK8861.3b05c9c5.0
Via: SIP/2.0/UDP 212.227.15.197;branch=z9hG4bKca1305302a7d41c334b7e10607ff0942
Via: SIP/2.0/UDP 217.188.44.231;branch=z9hG4bK8861.ec83b004.0
Via: SIP/2.0/UDP 1und1-1.sip.mgc.voip.telefonica.de:5060 ;received=195.71.9.100;branch=z9hG4bKterm-1e5356-
      +491712345678-+49601234567
From: +491712345678 <sip:+491712345678@1und1-1.sip.mgc.voip.telefonica.de;user=phone>;tag=1168841412
To: +49601234567 <sip:+49601234567@1und1.interconnect.sip.voip.telefonica.de;user=phone>
Call-ID: 7dbde21a-6a92b0f7-73ae04b2-b982@subscriber1.interconnect.mgc.voip.telefonica.de
CSeq: 1 INVITE
Supported: timer
Session Expires: 1800
Min-SE: 1800
Contact:  <sip:+491712345678@1und1-1.sip.mgc.voip.telefonica.de:5060>
Allow: INVITE,ACK,PRACK,SUBSCRIBE,BYE,CANCEL,NOTIFY,INFO,REFER,UPDATE
Max-Forwards: 6
Content-Type: application/sdp
Content-Length: 618
Diversion: <sip:49601234567@sip2.schlund.de;user=phone>;reason=additional

v=0
o-- 1710955 0 IN IP4 62.53.226.3
s=Cisco SDP 0
c=IN IP4 62.53.226.3
t=0 0
m=audio 18324 RTP/AVP 8 0 99 102 2 103 4 104 105 106 107 18 0 125 101 100
a-rtpmap:99 G726-16/8000
a=rtpmap:102 G726-24/8000
a=rtpmap:103 G7231-H/8000
a=rtpmap:104 G7231-L/8000
a=rtpmap:105 G729b/8000
a=rtpmap:106 G7231a-H/8000
a=rtpmap:107 G7231a-L/8000
a=rtpmap:125 GnX64/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:100 X-NSE/8000
a-fmtp:100 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=x-cpar: a=fmtp:100 192-194,200-202
a-X-cap: 2 image udptl t38
```

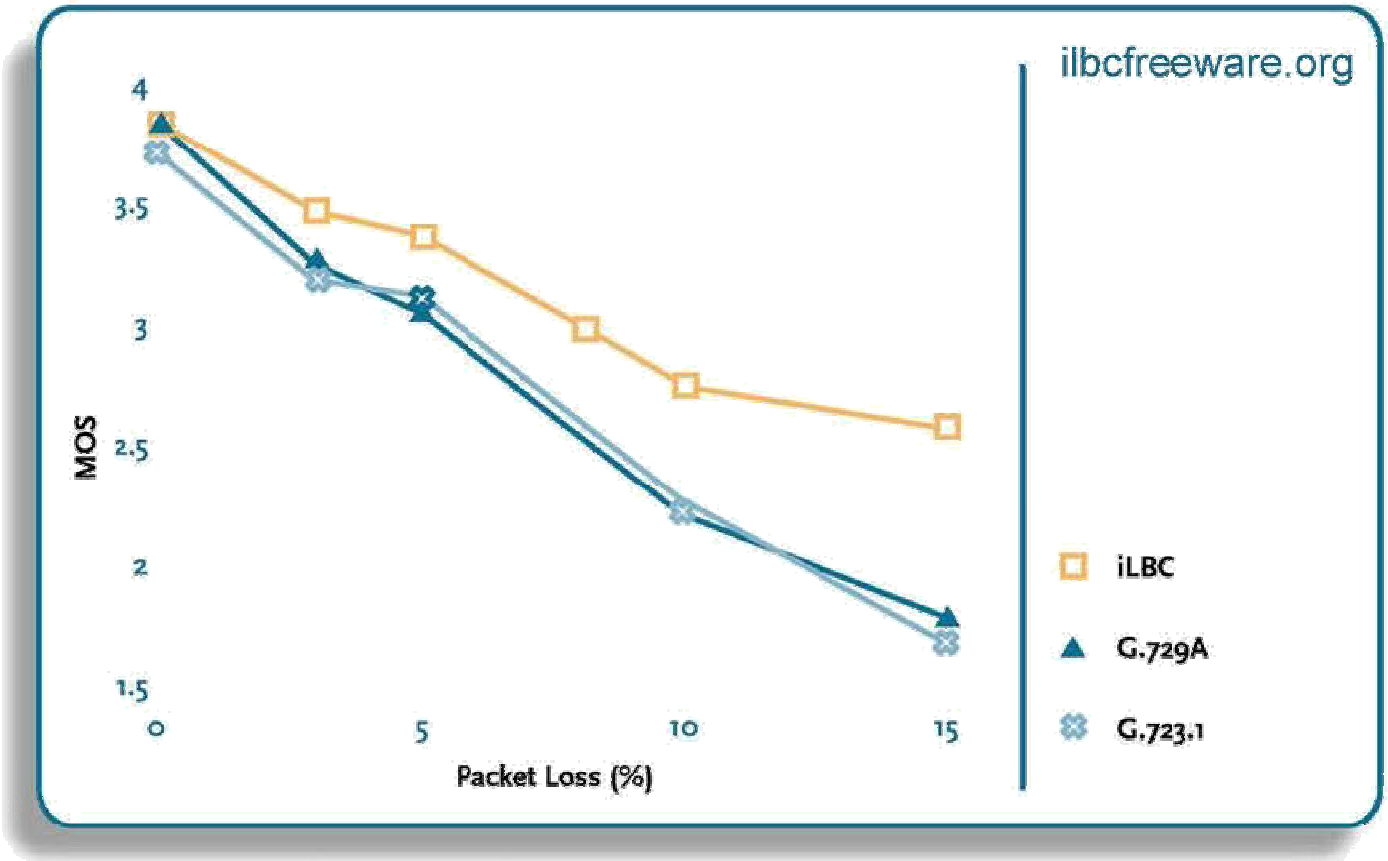
SIP-Header

SDP-Body

Salient characteristics of VoIP

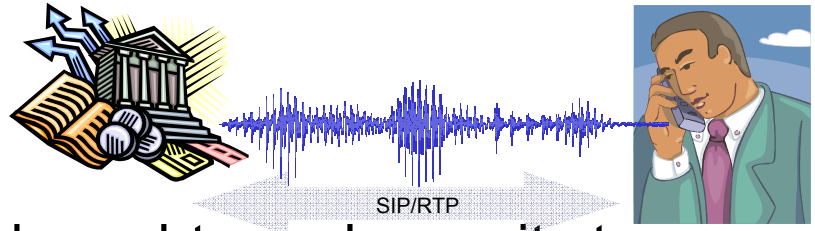
- As a communication channel, VoIP has rather specific features
- Telephony is **bidirectional** and **interactive**.
  - The medium consists in **linearly time-based full duplex channels**
  - Interactivity allows to make further enquiries in case of insufficient understanding
- VoIP chops communication into small pieces - packets, and transmits them independently
  - Packet loss is tolerated, methods to conceal it are standard and rather effective
  - Jitter (loss of temporal order) is common
  - Latencies >150ms are intolerable
- Typical: G.711 codec produces 64kbit/s, RTP packets come with 160 bytes of payload ~ 20ms

Packet loss vs. quality



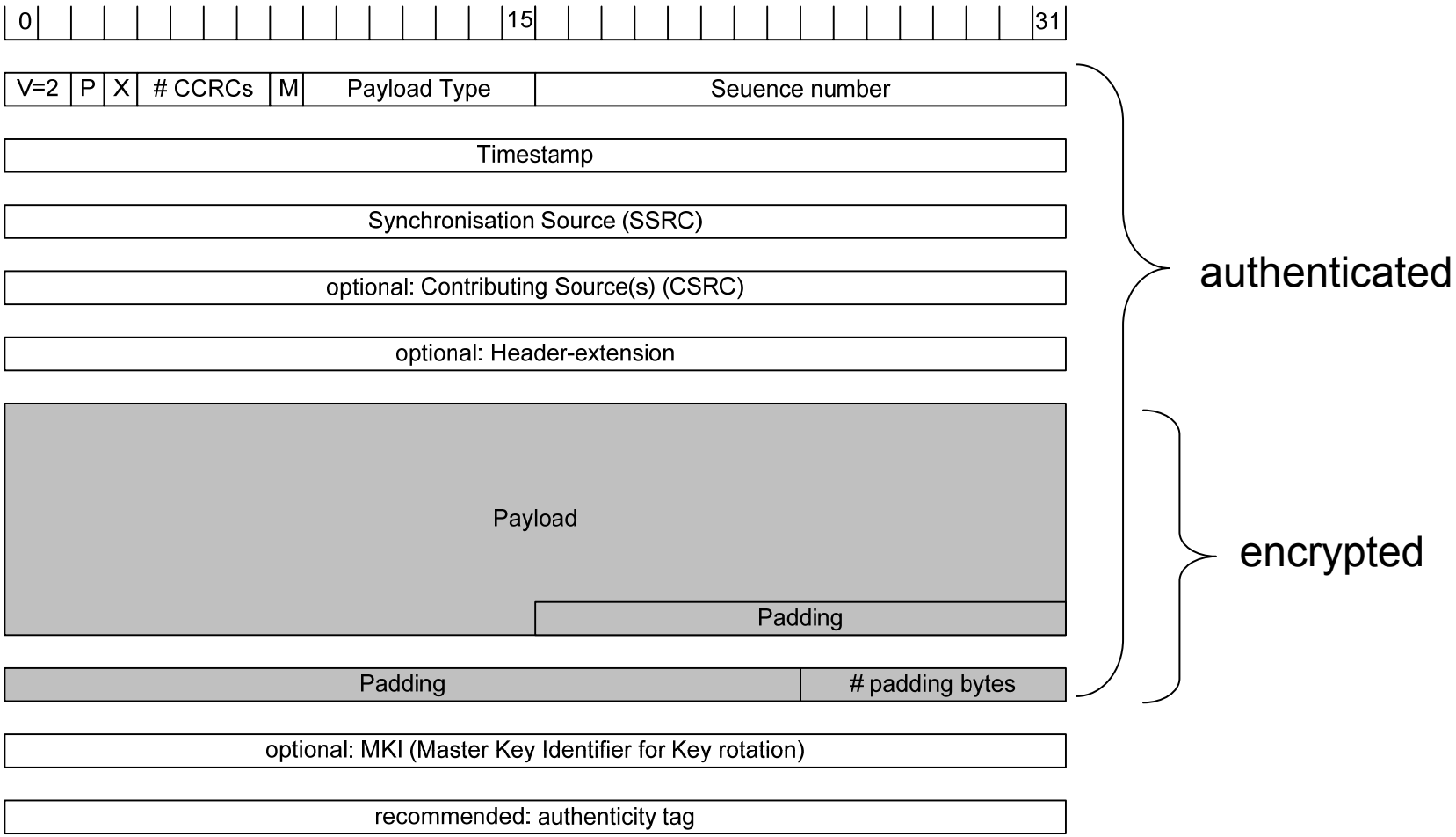
The tests were performed by Dynstat, Inc., an independent test laboratory.
Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

VoIP Security



- Basic issues are tackled
  - Protocols like SRTP can provide end-to-end security to phone calls, making them independent from the security of the transport medium. SPIT and DoS are future threats
  - Spam over Internet Telephony (SPIT) considered a major threat – approaches are Gatekeepers, CAPTCHAs, IDS, …
  - ITU Recommendations for Secure Telecommunications
- „Application-level security?“
  - Verbal communication is traditionally bestowed with a high level of trustworthiness
  - Is the integrity of VoIP communication as high as the inter-personal character of conversations suggests?
  - Recent court cases (contact centre v customer, digital recording accepted as evidence) suggest there might be a problem

SRTP – confidentiality and authenticity on packet level

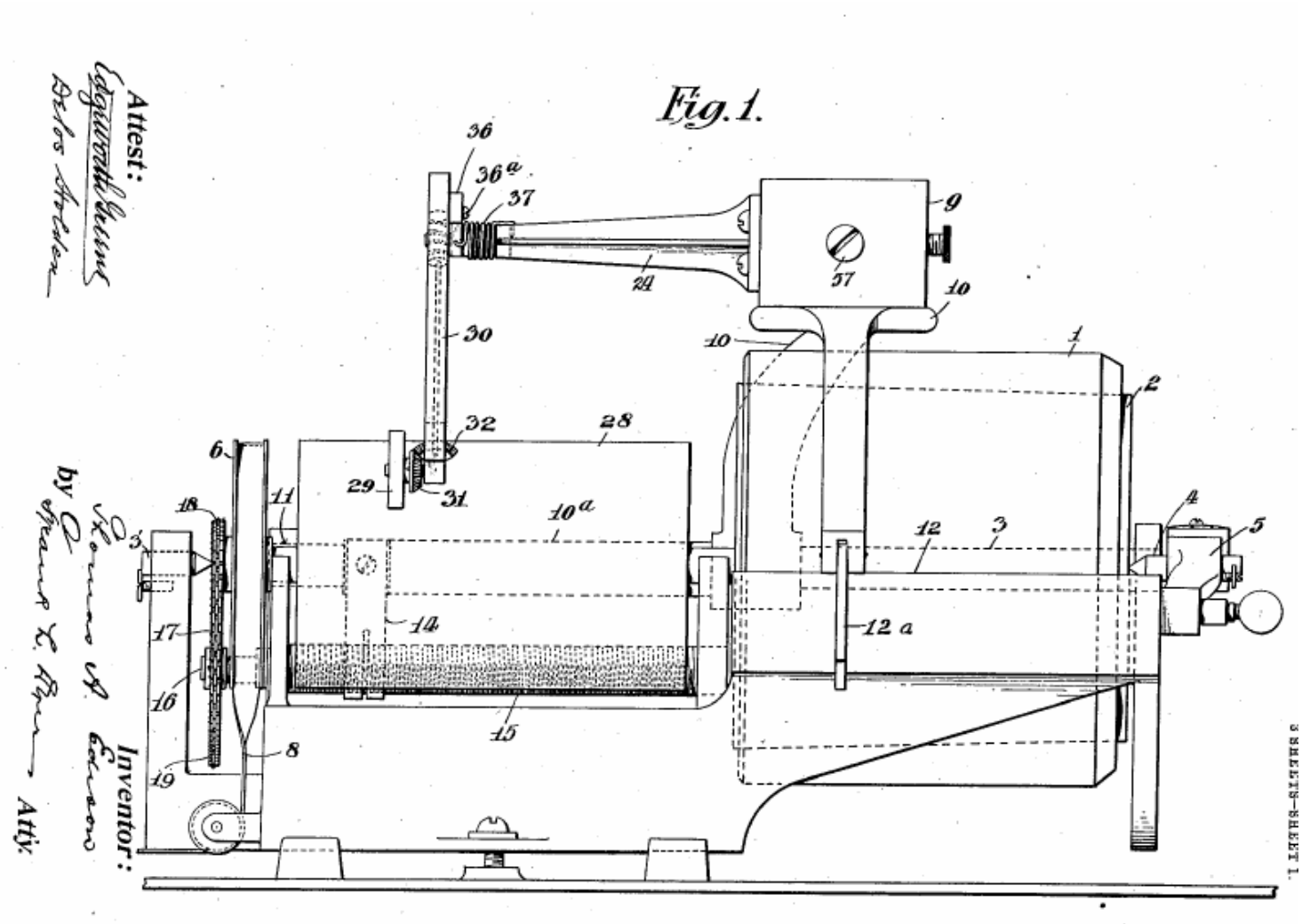


Authentication by HMACs (symmetric) and only on packet level

Non-repudiation for VoIP

- Goal: Non-repudiation of conversations by caller and callee, for speech over packet-oriented, digital channels, and in particular for VoIP conversations
- Providing tenable evidence of
  - Contents of a call
  - Identity of caller and callee
  - Ancillary information (forensic), time & date, …
- For electronic documents, this is usually done by electronic signatures

Trivia: What does this machine do?



The long history of voice non-repudiation

UNITED STATES PATENT OFFICE.

THOMAS A. EDISON, OF LLEWELLYN PARK, ORANGE, NEW JERSEY, ASSIGNOR TO THOMAS A. EDISON, INCORPORATED, OF WEST ORANGE, NEW JERSEY, A CORPORATION OF NEW JERSEY.

RECORDING-TELEPHONE.

1,012,250. Specification of Letters Patent. Patented Dec. 19, 1911.

Application filed September 15, 1905. Serial No. 278,549.

To all whom it may concern:

Be it known that I, THOMAS ALVA EDISON, a citizen of the United States, residing at Llewellyn Park, Orange, in the county of Essex and State of New Jersey, have invented certain new and useful Improvements in Recording-Telephones, of which the following is a description.

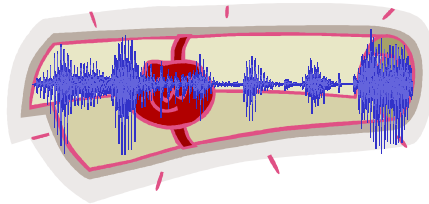
My invention relates to telephones and has for its object the provision of means whereby the electrical vibrations or undulations which are received over the line may be recorded phonographically, whereby a record is formed which may be used in any ordinary phonograph, and the message repeated at any future time.

telephonic receiver and a portion of the mechanism for driving the friction wheel; Fig. 4 is a section on line 4—4 of Fig. 3; and shows also the electrical connections.

In all the above views corresponding parts are designated by the same reference numerals.

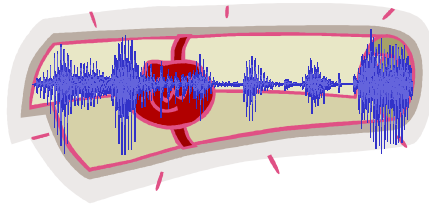
The recording surface may be a cylinder 1 of suitable material for receiving a phonographic record and the mechanism for supporting and rotating said cylinder may be similar to the parts of an ordinary phonograph comprising a tapered mandrel 2 on which the cylinder 1 is held by frictional engagement and carried by a shaft 3 supported at its end by pivot pins 3′ and 4

Requirements for Non-repudiation 1/2



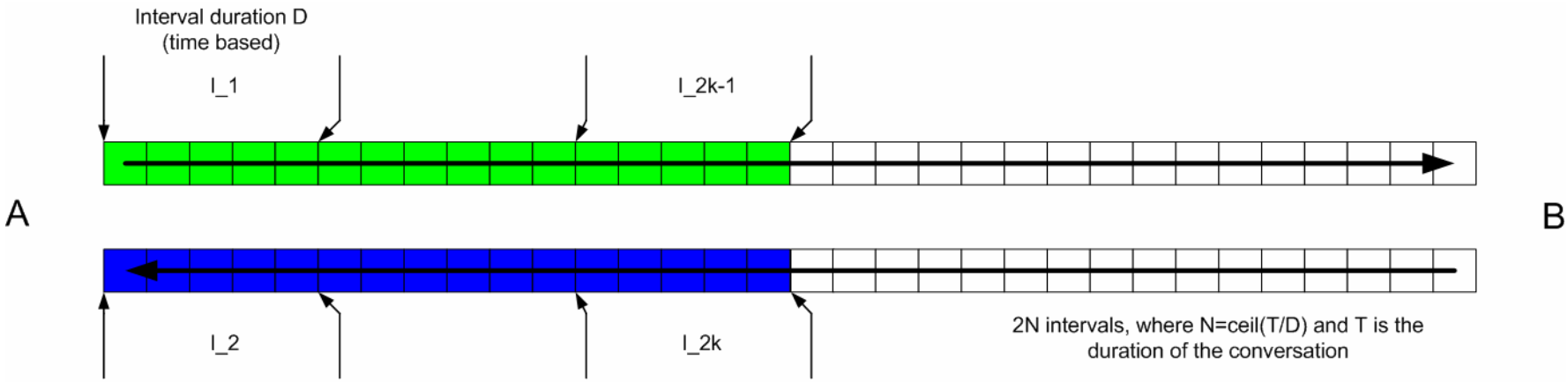
- Protection Target Congruence
  - Meaning can vary between sender and receiver
  - Electronic documents: „What You See is What You Sign“ tacitly assumes all parties „see“ the same
  - The receiver‘s understanding is essential
  - What is Heard is What is Signed
  - Communication partners need to agree on what was heard
- Requirements
  - 1.1 Integrity: The complete conversation and ist atomic parts must be secured
  - 1.2 Treatment of losses: a secure detection of losses enabling a proper handling on the application level as well as a later inspection.
  - 1.3 User interaction policies: e.g. notify users of quality under-runs, enforce breaks or repeats

Requirements for Non-repudiation 2/2



- Protection Target Cohesion
  - protection and preservation of the sequence the information flows in all directions of the channel
  - temporal reference frame of a conversation can be meaningful – fix conversation in absolute time
- Requirements
  - 2.1 Start times of conversations must be determined and recorded. This is analogous to the signing time of documents (the assignment of which is a requirement for qualified signatures according to the EU Signature Directive).
  - 2.2 Temporal sequencing within conversations must be protected and related to the reference time frame
  - 2.3 Continual authentication of communication devices and if possible even communication partners is necessary, e.g., to prevent hijacking
  - 2.4 Determined break points must allow for non-repudiation of conversations until they are terminated intentionally or inadvertently.

The interval chaining method



A signs a conversation with B, assuming no packet loss

$$\mathrm{Sec}_I: M_I \stackrel{\text{def}}{=} (D, \mathsf{SIP\_Data}, \mathsf{Auth\_Data}, \mathsf{nonce}, \ldots) \longrightarrow B;$$
$$S_0 \stackrel{\text{def}}{=} \left((M_I)_A\right)_{TS} \longrightarrow B;$$
$$\mathrm{Sec}_l: S_l \stackrel{\text{def}}{=} (I_l\ S_{l-1})_A \longrightarrow B; \quad l = 1, \ldots, 2N$$
$$\mathrm{Sec}_F: M_F \stackrel{\text{def}}{=} (\mathsf{termination\_condition}, \ldots) \longrightarrow B;$$
$$S_F \stackrel{\text{def}}{=} \left((M_F, S_{2N})_A\right)_{TS} \longrightarrow B;$$

$(\cdot)_X \stackrel{\text{def}}{=} \mathrm{Priv}_X(h(\cdot))$

Denotes the signing of data by entity X

Catering for packet loss by reporting of actually received packets

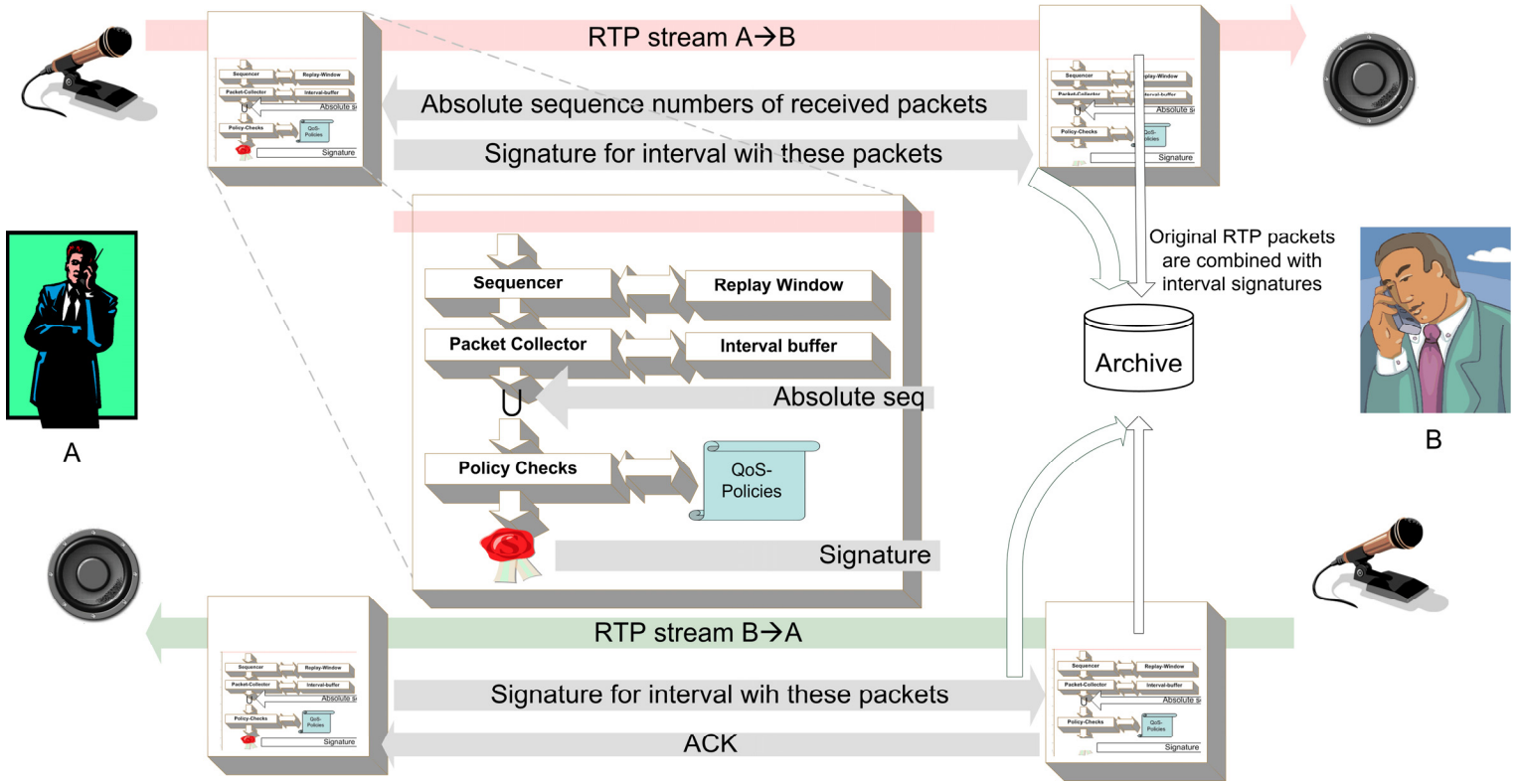
- List of packets actually received by B in interval I $\delta_l \subset \{1, \ldots, K_l\}$
- Reduced packet set in interval I $I'_l \stackrel{\text{def}}{=} (p_{l,j})_{j \in \delta_l}$
- For direction A to B, security procedure is modified as follows

Sec'_{2k-1} : repeat
  repeat
    interval_termination $\longrightarrow B$;
  until $\delta_{2k-1} \longrightarrow A$;
until $S_{2k-1} \stackrel{\text{def}}{=} (I'_{2k-1}, S_{2k-2})_A \longrightarrow B$;

- For direction B to A

Sec'_{2k} : repeat
  $S_{2k} \stackrel{\text{def}}{=} (I'_{2k}, S_{2k-1})_A \longrightarrow B$;
  $\delta_{2k} \longrightarrow B$;
until success;

Unidirectional signing procedure and architecture

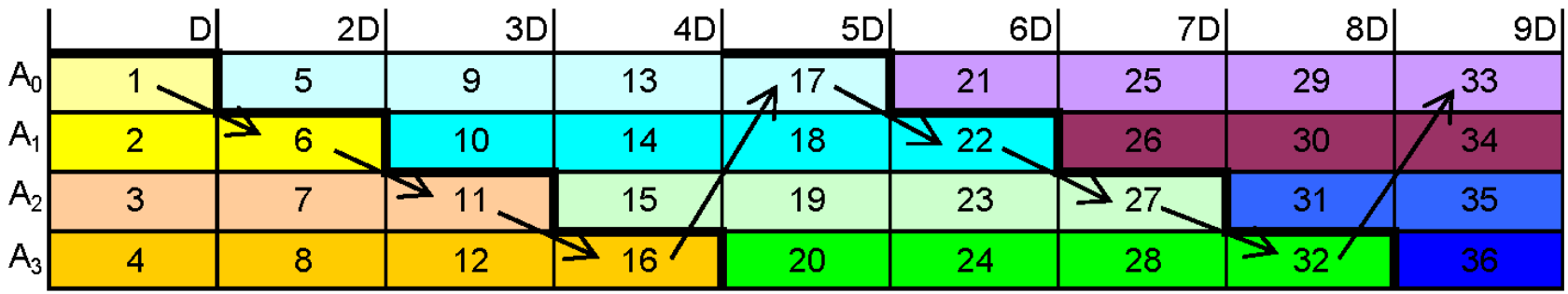


The original RTP-stream is forwarded to the recipient in real time without noticeable delay.
The **sequencer** extends the truncated 16Bit-sequence number of RTP-packets to 64 Bit, absolute sequence numbers starting with zero.
A **replay window** as defined in annex A of the SRTP-RFC is used to detect duplicate packets.
The **packet-collector** collects all sent or received packets and sorts them by their absolute sequence number. It buffers all packets belonging to the current interval. This has much less and only static memory requirements than storing the complete call for later signature.

For the channel A to B: A gets from B the **list with packet numbers that B actually received**. A as the sender of these packets was able to collect them all. A then discards the packets that B did not receive.
After that **QoS-policies** can be applied: If B lost to many packets, the call becomes ambiguous and A may terminate the call or take other measures.
A builds the **interval signature** package with metadata and the hashes of the contained RTP-packets and sends this to B. The full RTP-packets need not be send again over the wire thus resulting in an efficient implementation.

Multi-lateral signatures

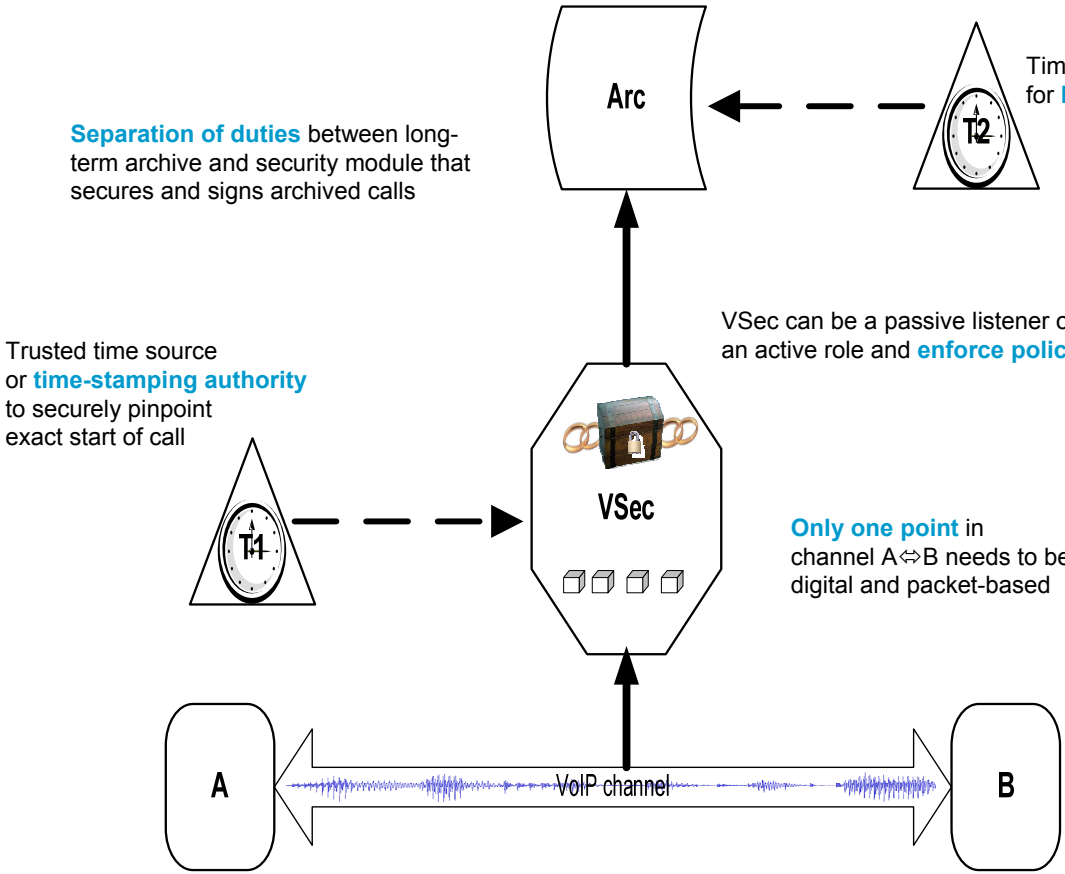
- Combine interval chaining with round-robin scheme
- A participant not carrying the token waits and buffers packets sent by himself
- When participant A_m carries the token he waits D, and buffers packets sent by himself
- A_m terminates an interval, secures it, and foregoing intervals not yet secured and broadcasts the security package
- Packet loss: include hashes of packets received by at least one participant in the security package
- Signatures are verifiable for all receivers
- The security package is used by A_m+1 to continue the chaining

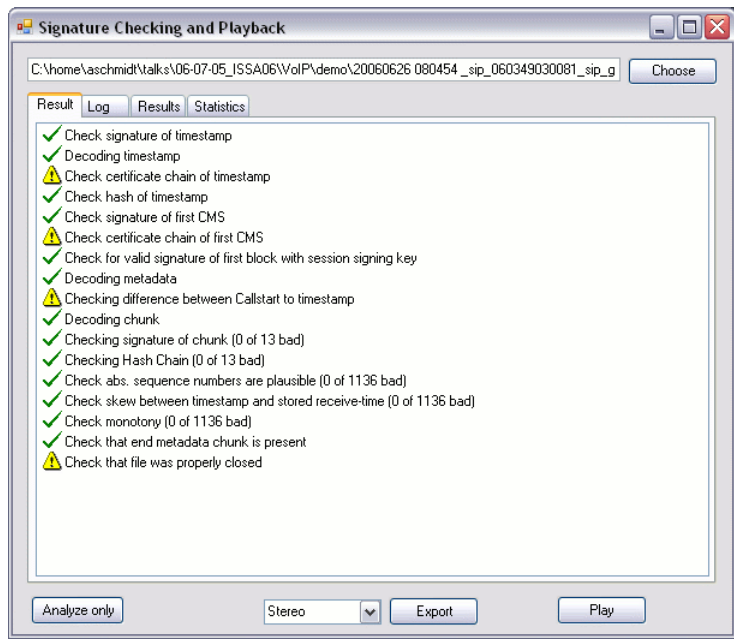


Auditable information from the interval chaining method

Auditable item	Req.	Protection target	Verifies/indicates	When applicable
Initial time stamp	2.1	Cohesion	Start time	Always
Initial signature & certificate	2.3	Cohesion	Identity of signer	Always
Interval Chaining	2.2, 1.1	Cohesion	Interval integr. & order	Always
Packet loss in intervals	1.2, 2.4	Congruence	QoS, understandability	Always
Monotonic increase of RTP-sequence numbers	1.1, 2.2	Integrity & cohesion	RTP-stream plausibility	Always
Relative drift of RTP-time marks against system time	2.2	Cohesion	RTP-stream plausibility	During convers.
Relative drift of RTP-time marks against $\lfloor l/2 \rfloor \cdot D$	2.2	Cohesion	Packet & stream plausibility	Ex post
No overlaps of RTP-time marks at interval boundaries	2.2	Cohesion	Packet & stream plausibility	Always
Replay-window		Integrity	Uniqueness of recorded audio stream	Always
Final time stamp	2.2	Cohesion	Conversation duration	Ex post
Forensic analysis of recorded conversation		(Semantic) authenticity	Speaker identity, mood, lying, stress, etc.	Ex post, forensic

Application: Self-signed VoIP Archive





Main Benefit: high accountability by separation of duties and resulting administrator security

Conclusions

Interval chaining – VoIPS – achieves non-repudiation for VoIP conversations with the salient features

- Saves investment and operational cost
  - Modular architectures
  - Scalability
  - Pluggable into existing telecommunication infrastructures with minimal effort
  - Operates without interference (causes no latency)
- Highest security for reasonable investment
  - Complete audit trail for digital voice communication
  - Administrator-proof security essential for accountability
  - Cost-efficient backup and failover solutions
- Enabling new forms of non-repudiation in co-operation
  - Continual caller authentication
  - Verbal contracts between unacquainted persons
  - Non-repudiation in (multi-media) conferences
- Potential users:
  - Call centres, small businesses, administrations, financial institutions, mobile users, …

Patent pending